Avaya B179 SIP Conference Phone

Installation and Administration Guide

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ABOUT THIS DOCUMENT

This document includes setup, registration of accounts and configuration of Avaya B179 Conference Phone in Communication Manager, Session Manager, and CS-1000 Server environments.

For information about setting up B179 in IP Office, see Installing and Administering the IP Office B179 SIP Conference Phone.

The use of the conference phone is described in the Avaya B179 SIP Conference Phone - Quick Reference Guide (16-603916) and the Avaya B179 SIP Conference Phone - User Guide (16-603918). The latest version of all documentation can be downloaded from support.avaya.com.

Please note that there are also supporting Application Notes describing the steps to configure the Avaya B179 SIP Conference Phone to work with certain systems and also how to configure the systems (eg. administer SIP extensions).

RELATED RESOURCES

Title	Description
Installing and Administering the IP Office B179 SIP Conference Phone	Describes the procedures to install and administer B179 SIP Conference Phones in IP Office.
Administering Avaya Aura® System Manager	Describes the procedure to administer Avaya Aura® System Manager

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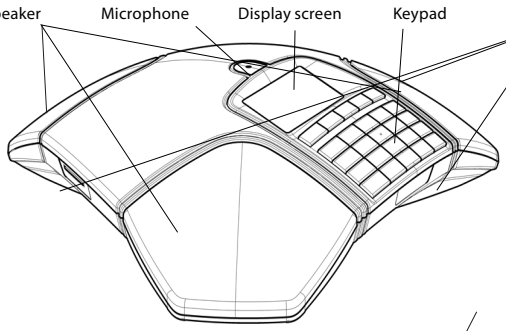
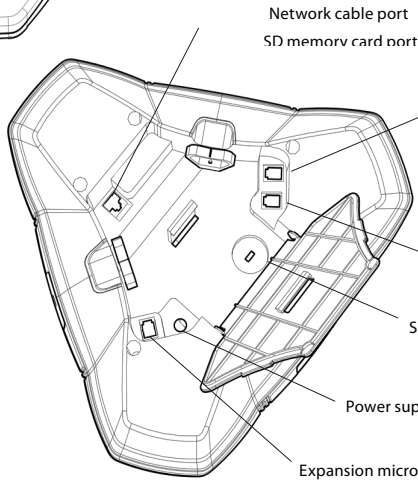
DESCRIPTION



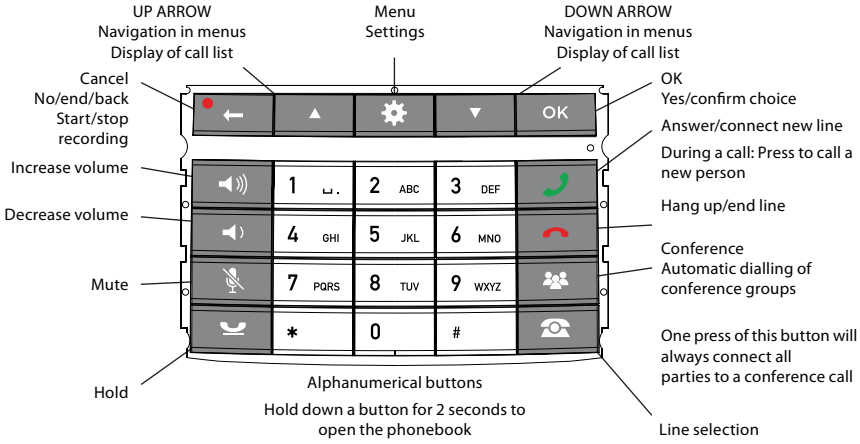
LEDs

Flashing blue	Incoming call
Steady blue light	Call in progress
Flashing red	On hold, microphone and speakers turned off
Steady red light	Mute, microphone turned off



Maintenance

Clean the equipment with a soft, dry cloth. Never use liquids.

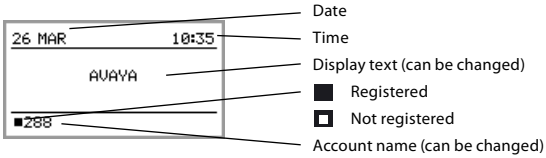


ⓘ Some Avaya B179 have a different keypad with other symbols. This does not affect the functions of the buttons.

DISPLAY SCREEN

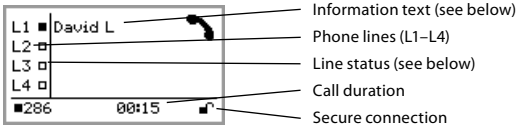
On Hook

Press to display this screen.



Off Hook

Press to display this screen.



Line status:

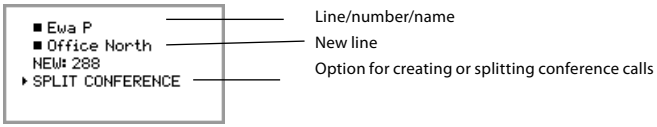
- Line free (Before account name – telephone not registered)
- Line connected (Before account name – telephone registered)

Information text displays one of the following:

- Number or name of each phone line
  (The name will be displayed if a number is in the phone book)
- Explanation of what you should do (For example, ENTER NUMBER)
- Status (For example Hold when you place all calls on hold)

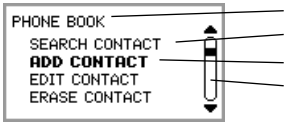
Line menu

Press to switch to and from this menu.

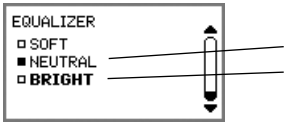


Menu

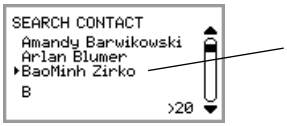
Press ✹ to switch to and from a menu.



Current menu
Submenu
Marked option – open by pressing OK button
Scrolling list
(indication of where the marked option is in the list or menu)



List of setting options:
Existing settings
Marked option – select by pressing OK button



List of names:
Marked name – select by pressing OK button

NAVIGATION AND SELECTION IN MENUS

⇨ Press ✹.

⇨ Select the option you want from the menu using the arrow buttons.

⇨ Confirm by pressing OK to select the marked option.

⇨ Cancel the setting or go back one level in the menu by pressing ■ ←.

⇨ Quit the menu by pressing ✹ again.

ⓘ Note that after you have made changes to a setting, you must press OK to activate the setting.

It is possible to open a menu option directly by pressing the number button that corresponds to the position of the option in the menu (For example, 2 to open PHONE BOOK and then 3 to select EDIT CONTACT).

Writing style in instructions

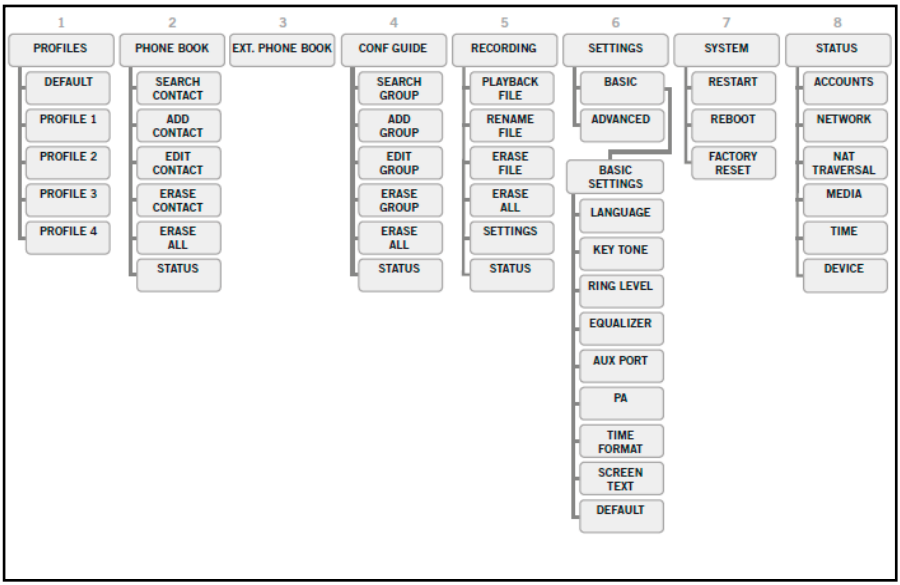
In the instructions, ✹ > SETTINGS (6) means you should:

⇨ Press ✹.

⇨ Mark the SETTINGS option using the arrow buttons and confirm by pressing OK to open the menu (or press button number 6).

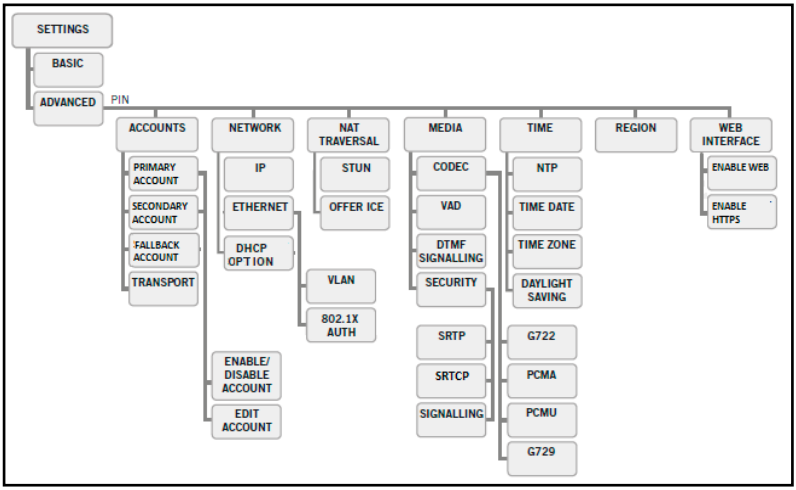
Correspondingly, Phone book > Conference Guide in the web interface means you should select Menu Phone book and the Conference Guide tab.

Menu tree



Menu tree, advanced settings

The advanced settings are protected by administrator's PIN code. The default value is 1234.



The simplest way to make settings and edit contacts is using a PC and the Avaya B179 web interface.

USING THE WEB INTERFACE

You can use the web browser of a PC connected to the same network to manage contacts, conference groups and settings in the Avaya B179. Users can import and export contacts and conference groups, name user profiles, and change PIN codes by using the web interface.

All settings on the Avaya B179 can also be managed via the web interface. You can also view logs, update software, and create a configuration file to use for other phones.

ⓘ The values set by using the web interface overrides the local settings on the phone and the values set by using the settings file.

The default setting for the PIN code is 0000 for the user account (Default, Profile 1, Profile 2, Profile 3 and Profile 4) and 1234 for the administrator's account (Admin). You must change the PIN codes in order to protect the settings. The code may consist a maximum of eight digits. The administrator can always view and change the PIN codes to the user accounts. The administrator's PIN code can only be reset with a complete reset to factory settings.

ⓘ The existing value of PIN codes are retained after firmware upgrade.

Disabling the web interface

You can disable the web interface on the B179 conference phones to restrict network-based access to the conference phones. When you configure the phone to disable the web interface, the HTTP server application on the phone is disabled, and the users cannot access the web interface.

Note:

You can enable the web interface on a phone from the phone UI. The local web interface can be used for debugging, but the phone cannot be accessed over the network.

⇨ In the `www` tag in the global configuration file, set the `</enable>` tag to false.

Syntax:

```
<www>
<enable>false</enable>
</www>
```

⇨ Set the Web Interface option in the phone UI to DISABLE WEB.

⇨ Press ⚙ and select the sub menu > ADVANCED (requires Admin password) > WEB INTERFACE > DISABLE WEB.

You can use the configuration file of the phone to edit the phone settings.

For more information, see topic Using a configuration file.

INSTALLATION

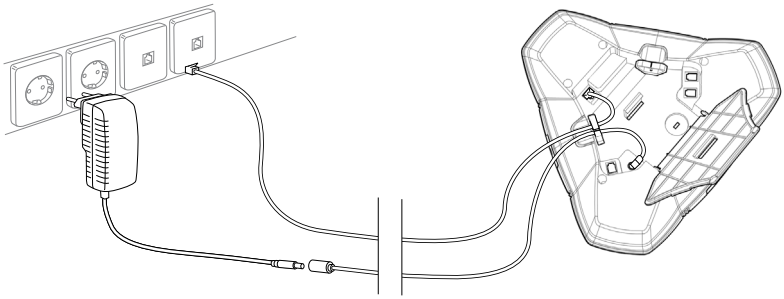
The printed Installation Guide provides brief and simplified installation instructions.

CONNECTING

⇨ Connect the Avaya B179 to the network as illustrated below.

⇨ Plug the Avaya B179 into the mains using the power adapter as illustrated below.

ⓘ The Avaya B179 can be driven directly from the network (Power over Ethernet, Class III) if the network supports this.



⇨ Place the conference phone in the middle of the table.

The Avaya B179 must obtain a network address and be registered in a SIP PBX before it can be used. The easiest way to register an account and make the settings in the Avaya B179 is using a computer connected to the same network and via the integrated web server.

OBTAINING A NETWORK ADDRESS

Connecting to a network with DHCP

After the B179 Conference Phone connects to a DHCP network, the phone requests for the network parameters.

To enable DHCP:

⇨ In the web UI, go to Settings > Network. Set the DHCP option to Enabled.

⇨ In the phone UI, Press ✱ and select Settings > Advanced (requires the Admin password) > Network > IP > DHCP. Set the DHCP option to On.

DHCP Site-specific option (SSON)

You can assign the values of site-specific configuration parameters by using DHCP SSON.

You can enable SSON by any of the following methods.

⇨ Open the configuration file in an editor. In the <dev_mgmt> tag, set the value of the <dhcp_option> parameter.

⇨ In the web interface of the phone, go to Settings > Provisioning > DHCP Option.

⇨ In the phone UI, Press ✱ and select Settings > Advanced (requires the Admin password) > Network > DHCP Option.

ⓘ The default value for DHCP SSON is 242. B179 supports SSON from 128 to 254, and option 60 for backward compatibility.

Connecting to a network with static IP addresses

You need the IP address, host name, domain, netmask, gateway, DNS 1, and DNS 2. The host name can be set freely. The domain and secondary DNS can be left blank.

⇨ Press ✱ and select SETTINGS > ADVANCED (6,2).

⇨ Enter the PIN code.

ⓘ The default code is 1234.

⇨ Select NETWORK (2)

⇨ Select IP.

⇨ Select STATIC IP.

⇨ Enter values for the IP ADDRESS.

ⓘ Enter three digits (begin with 0 if necessary), press OK, enter three digits, and so on.

⇨ Enter HOST NAME
Default is avaya.

⇨ Enter DOMAIN

⇨ Enter NETMASK

⇨ Enter GATEWAY

⇨ Enter DNS 1

⇨ Enter DNS 2
The display shows DONE.

Checking IP address

⇨ Press ✱ and select the sub menu STATUS > NETWORK (8,2).

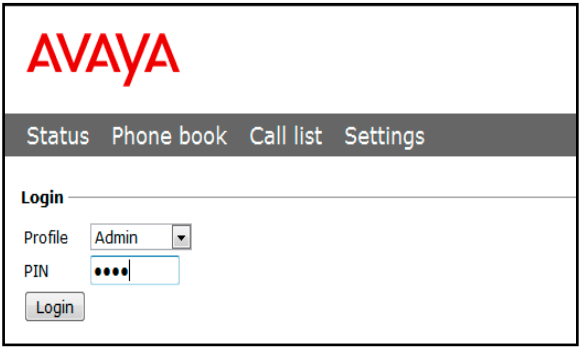
⇨ Check the conference phone's network address under the heading IP ADDRESS.

ⓘ Use this address to log into the web server in the conference phone.

Login

⇨ Log into the web server in Avaya B179 by entering the phone's network address in your computer's web browser.

⇨ Select Admin as Profile and enter your PIN. The default PIN for Admin is 1234.



SOFTWARE UPGRADE AND BASIC SETTINGS

The following settings should be done during installation.

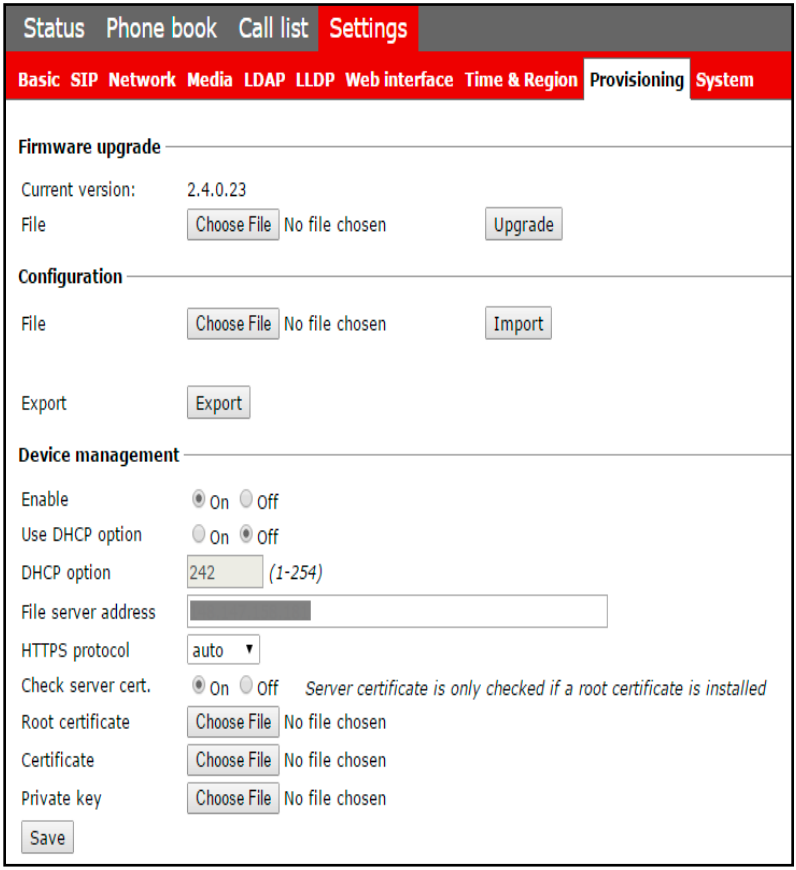
ⓘ See Device management, if you are responsible for installing or upgrading many phones.

Note that all settings on the Basic tab also affect the user profile Default. Other user profiles can be changed individually. The settings on the Basic tab, except the name and PIN for Admin, can be modified by any user. Other settings require a login as Admin.

Upgrade software

See the heading “PROVISIONING – UPGRADE AND CONFIGURATION” for a detailed description and upgrading options.

⇨ Select Settings > Provisioning.



⇨ Compare the latest version with the current version (shown on the web page).

⇨ If you want to upgrade, select the desired version in the list box and click on Upgrade.
The browser window and the display on the Avaya B179 shows that the upgrade has begun.

ⓘ The download and installation can take several minutes. Do not interrupt the upgrade and do not disconnect plugs to the Avaya B179 during the upgrade. Interrupting the upgrade may render the conference phone inoperable.

⇨ When installation is complete, the text “Upgrade Complete. The unit will be rebooted.” is shown in your browser, and after a while you hear the Avaya music signature, which indicates that the conference phone has started.

Setting time and region

⇨ Select Settings > Time & Region.

⇨ Select the time zone and, if you wish, correction for DST (Daylight saving).

ⓘ You can set the time and date manually or choose a different time server.

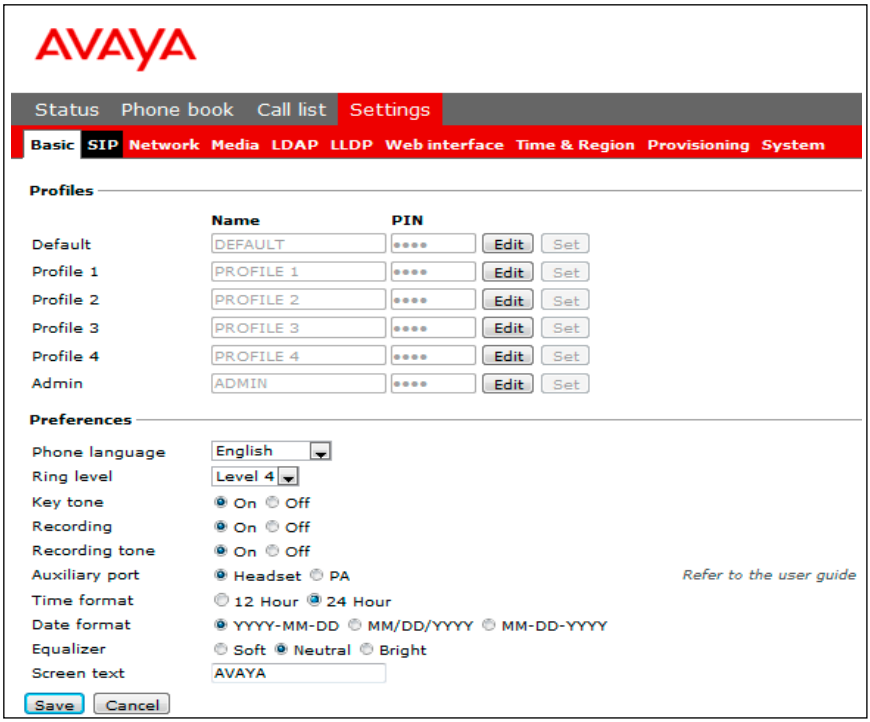
⇨ Select the region where you are.

ⓘ This setting affects the frequency and duration of the signaling tones (ring signal, busy tone, etc).

⇨ Save the setting.
The Avaya B179 restarts to apply the new settings.

Changing the language

⇨ Select Settings > Basic.



⇨ Select the desired language in the list box after Language and save the setting. The B179 Phone supports the following languages:

English

Swedish

Danish

Norwegian

Finnish

Italian

German

Spanish

Portuguese (Br)

Portuguese (Eu)

French

Dutch

Cyrillic

Polish

Turkish

Greek

Simplified Chinese

Japanese

Korean

ⓘ Note, selecting a language only affects the phone language, not the language on the web interface.

Changing the PIN

You must change the PIN code for Admin from the default setting to protect the settings. Make a note of the new PIN code and keep it in a safe place. The administrator's PIN code can only be reset by a full factory reset!

⇨ Select Settings > Basic and click the Edit button on the Admin line.

⇨ Enter a new PIN.

ⓘ The PIN code may consist a maximum of 8 digits.

⇨ Click on the Set and Save buttons.

REGISTERING AN ACCOUNT

The conference phone supports three accounts. The secondary and fallback accounts are automatically used if the phone fails to register to the main account. If the phone fails to register to the secondary or the fallback account, it tries to use the main account again.

To register your phone, you must have access to the account information and all necessary settings that the SIP PBX requires.

See topic Settings > SIP for a detailed description of all settings.

Procedure

⇨ Under Main account, Click Yes at Enable account.

⇨ Enter the account information you have received.

ⓘ Account name can be chosen freely and is the name or phone number you want to appear in the phone display.

⇨ Leave the default values if you have no other information.

⇨ Select a method of NAT traversal if you have received this information.

⇨ Select a different transport protocol if you have received this information. See page 28 about using a secure transport protocol.

⇨ Save the settings by clicking the Save button.

The Avaya B179 responds by showing REGISTERING. If registration is successful, your selected account name will appear at the bottom of the display screen next to a shaded square.

Media settings

⇨ Select a different codec priority, if you do not accept the default settings.

⇨ Select SRTP and SRTCP if you need a secure media protocol. Note that this also requires a corresponding transport setting on the SIP tab.

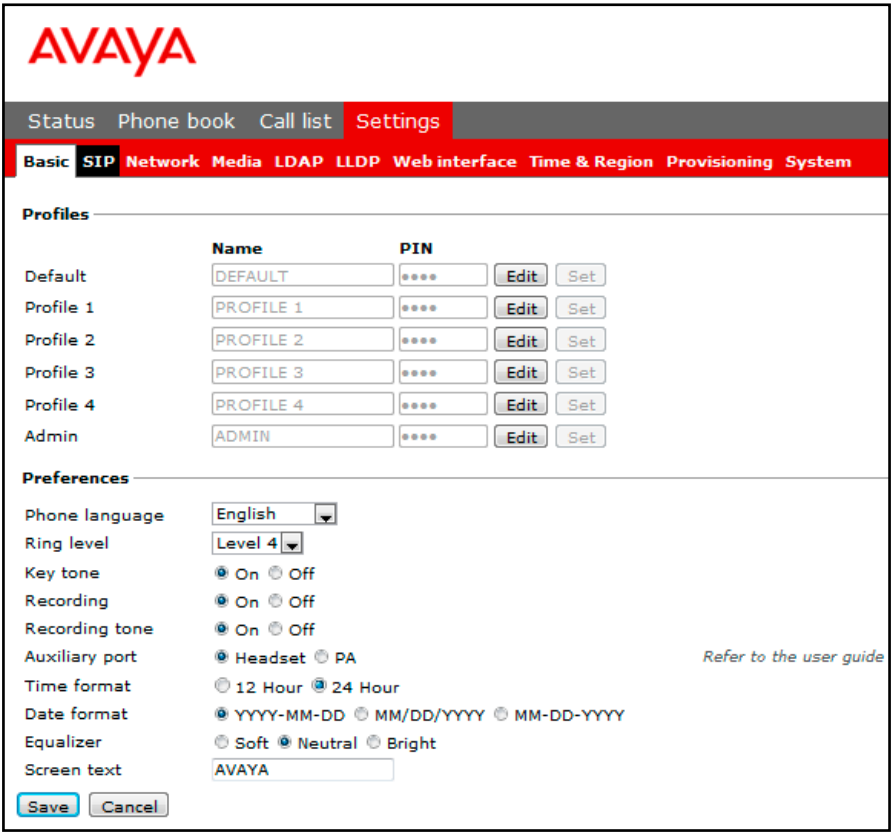
SETTINGS

You can use the web interface for the settings that can be done directly on the B179 Phone. See “NAVIGATION AND SELECTION IN MENUS” for using the menu system.

For safety reasons, recordings can only be managed directly on the Avaya B179. All other settings can be changed via the web interface. The web interface also allows you to import and export contacts and conference groups, rename user profiles and change PIN codes. As an administrator, you can also study logs, upgrade the software and create an XML based configuration file for easier management of a set of phones.

BASIC

⇨ Select Settings > Basic.



ⓘ To change the basic settings of a user profile, you must log in with that profile.

Profiles – edit name and PIN

You must change the PIN code from the default setting to protect the settings.

⇨ Select Settings > Basic and click the Edit button on the account you want to change.

⇨ Enter a new PIN code.

ⓘ The PIN code may consist of 8 digits.

ⓘ You can also choose to change the name of a user profile.

⇨ Click on the Set and Save buttons.

ⓘ Make a note of the new PIN code and keep it in a safe place.

ⓘ The administrator's PIN code can only be reset with a complete reset to factory settings!

Language

⇨ Select phone language using the list box and click on the Save button.

On phone: ✱ > SETTINGS > BASIC > LANGUAGE (6,1,1).

Ring level

There are six volume levels plus a silent mode. You will hear the ring tone for each level you select. If you select silent mode, only the blue LEDs on the phone flash when an incoming call is received.

⇨ Select level using the list box and click on the Save button.

On phone: ✱ > SETTINGS > BASIC > RING LEVEL (6,1,3).

Key tone

You can select whether or not you want a tone to be heard when you press a button.

⇨ Select On or Off and click on the Save button.

On phone: ✱ > SETTINGS > BASIC > KEY TONE (6,1,2).

Recording

It is possible to turn off the recording feature. This setting can only be done by the administrator and affects all profiles.

⇨ Select On or Off and click on the Save button.

For security reasons, recordings can only be managed directly on the SD card on Avaya B179.

ⓘ Do not remove the SD card during call recording or playback.

Recording tone

A short beep is heard every 20 seconds so that all the parties in the call know it is being recorded. This feature can be turned off.

⇨ Select On or Off and click on the Save button.

On phone: ✱ > RECORDING TONE > SETTINGS (5,5).

Settings when connecting external equipment (Aux)

The Avaya B179 can be connected to a wireless headset or an external PA system. An optional PA interface box is required for PA system connection.

⇨ Select the PA option to activate features for external microphone mixer and PA system.

ⓘ Do not select the PA option unless a PA system is connected. This option turns off the internal microphone and internal speakers as default. The HEADSET option may be selected whether or not a headset is connected.

Time format

⇨ Select 12 hour or 24 hour and click on the Save button.

On phone: ✱ > SETTINGS > BASIC > TIME FORMAT (6,1,7).

Date format

⇨ Select date format and click on the Save button.

On phone: ✱ > SETTINGS > BASIC > DATE FORMAT (6,1,8).

Equalizer

The sound reproduction can be adjusted to the required pitch (SOFT, NEUTRAL or BRIGHT).

⇨ Select Soft, Neutral or Bright and click on the Save button.

On phone: ✱ > SETTINGS > BASIC > EQUALIZER (6,1,4).

Screen text

The text on the display screen is shown when the Avaya B179 is in stand-by mode (on hook).

⇨ Enter your new text in the text box and click on the Save button.

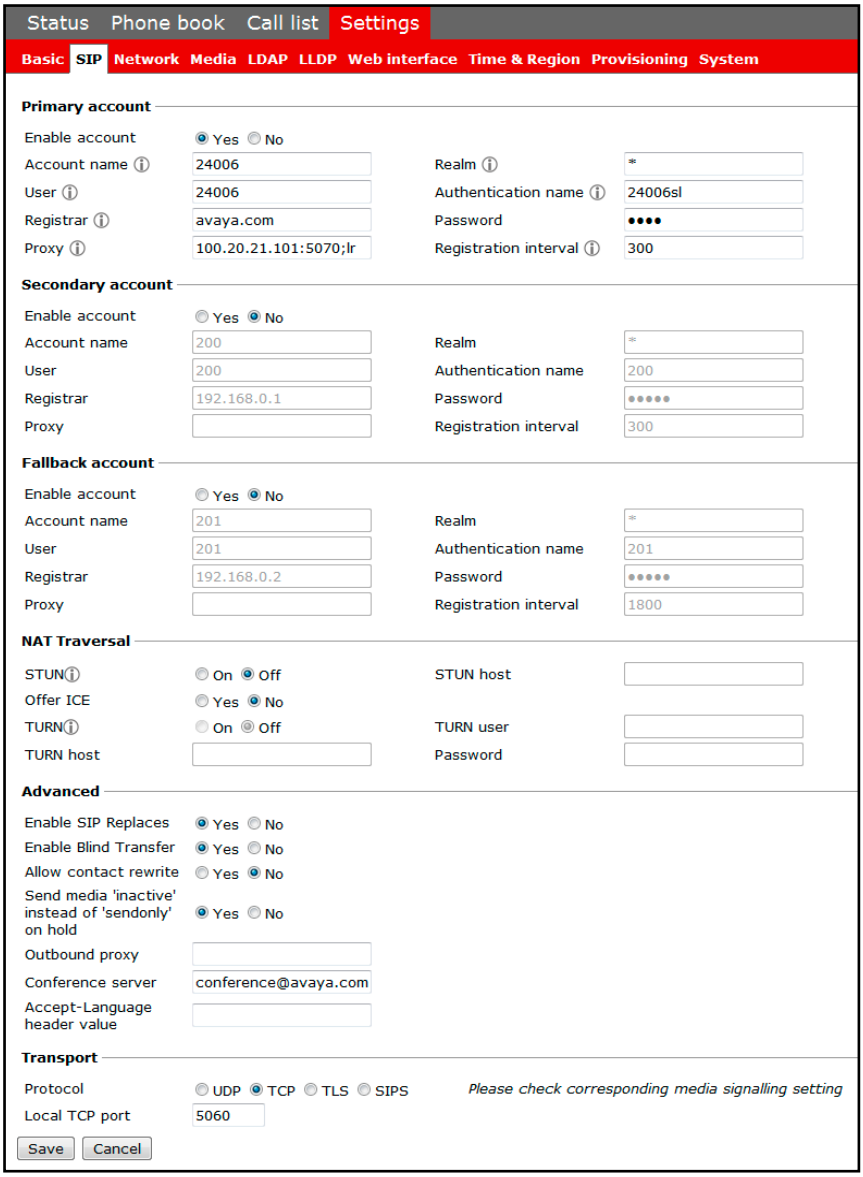
On phone: ✱ > SETTINGS > BASIC > SCREEN TEXT (6,1,9).

The table below lists the supported languages for screen text, and the allowed string length.

Language	String length allowed
English	11
Swedish	9
Danish	10
Norwegian	11
Finnish	12
Italian	13
German	11
Spanish	14
Portuguese (Br)	13
Portuguese (Eu)	14
French	11
Dutch	11
Cryllic	14
Polish	16
Turkish	11
Greek	14
Simplified Chinese	4
Japanese	9
Korean	6

SIP

⇨ Select Settings > SIP.



The conference phone supports three accounts. The secondary and fallback accounts are automatically used if the phone fails to register to the main account. If the phone fails to register to the secondary and the fallback accounts, it tries to use the main account again.

Main account, Secondary account, and Fallback account

Enable account	It is possible to store account information for future use, but temporarily disable it.
Account name	This is the name displayed on the screen. It can be set according to company standards.
User	The account (customer) name.
Registrar	Shall contain the IP address or the public name of the SIP server where the account is registered (e.g. 10.10.1.100 for a local SIP server or sip.company.net for a public VoIP service provider)
Proxy	Shall contain the proxy server used for Internet communication, if any. Can be left blank.
Realm	The protection domain where the SIP authentication (name and password) is valid. This is usually the same as the registrar. If marked with a "*", the phone will respond to any realm. If specified, the phone will only respond to the specific realm when asked for credentials.
Authentication name	The name used for the Realm authentication. This may be the same as the user name, but must be filled in.
Password	The password used for the Realm authentication.
Registration Interval	This is a request to the SIP server for when the registration should expire. Avaya B179 automatically renews the registration within the time interval if the phone is still on and connected to the server. The default value is 1800 seconds.

On phone: ⚙ > SETTINGS > ADVANCED > (PIN) > ACCOUNTS (6,2,1).

Nat traversal

NAT (Network Address Translation) is a firewall or router function that operates by rewriting the IP addresses in the IP headers as packets pass from one interface to the other. When a packet, for example, is sent from the inside, the source IP address and port are rewritten from the private IP address space into the address space on the outside (Internet).

NAT rewrites the addresses but leaves the packets themselves untouched. This kind of translation works fine for many protocols, but causes a lot of trouble for SIP packets that contain address information in their content (for example an INVITE request from one IP address to another).

NAT traversal solves this problem, providing a "view from the outside" that makes it

possible to replace the IP address in the SIP requests with the address shown on the other side of the firewall.

Note that in some cases NAT traversal is not necessary. Some public service providers of IP telephony keep track of the actual IP address used to register a phone, and the one used in the SIP requests from the same phone, and then replaces the addresses in the SIP messages.

STUN	STUN (Simple Traversal of UDP through NATs) is a protocol that assists devices behind a NAT firewall or router with their packet routing. STUN is commonly used in real-time voice, video, messaging, and other interactive IP communication applications. The protocol allows applications operating through a NAT to discover the presence and specific type of NAT and obtain the mapped (public) IP address (NAT address) and port number that the NAT has allocated for the application's User Datagram Protocol (UDP) connections to remote hosts. The protocol requires assistance from a 3rd-party network server (STUN server). STUN should be activated if an external SIP server cannot connect to the Avaya B179 behind a firewall NAT function and the SIP server supports STUN. A suitable STUN server is usually provided by the VoIP service provider. Note: STUN might also be referred to as Session Traversal Utilities for NAT.
STUN host	The IP address or public name of the STUN server.
Offer ICE	ICE (Interactive Connectivity Establishment), is a STUN addition that provides various techniques to allow SIP-based VoIP devices to successfully traverse the variety of firewalls that may exist between the devices. The protocol provides a mechanism for both endpoints to identify the most optimal path for the media traffic to follow.
TURN	TURN (Traversal Using Relay NAT) TURN is an extension of the STUN protocol that enables NAT traversal when both endpoints are behind symmetric NAT. With TURN, media traffic for the session will have to go to a relay server. Since relaying is expensive, in terms of bandwidth that must be provided by the provider, and additional delay for the media traffic, TURN is normally used as a last resort when endpoints cannot communicate directly.
TURN User	User authentication name on the TURN server.
TURN host	The IP address or public name of the TURN server.
Password	User authentication password on the TURN server.

On phone: ⚙ > SETTINGS > ADVANCED > (PIN) > NAT TRAVERSAL (6,2,3).

Advanced

Enable SIP Replaces	Default is Yes. Setting this option to No, will instruct the PBX not to use the SIP replace header. Some PBXes try to take over the bridging functionality from Avaya B179 using this command, which causes the calls to interrupt.
Enable Blind Transfer	Default is Yes. Setting this option to No, will disable the transfer function (✱ > TRANSFER) during a call. This may be used if the PBX does not support blind transfer.
Allow contact rewrite	Default is Yes. When enabled, the B179 will store the IP address from the response of the register request. If a change is detected, the phone will unregister the current sip URI (contact), and update the sip URI with the new address.
Send media 'inactive' instead of 'sendonly' on hold	Default is No. Must be set to Yes in a CS1000 server environment.
Outbound proxy	The IP address of the outbound proxy server.
Conference server	The IP address of the conference server that is currently in use.
Accept-Language header value	Indicates the preferred languages for reason phrases, session descriptions, or status responses carried as message bodies in the SIP response.

Transport

The transport setting only concerns the protocol to be used for SIP messages between the devices involved. These settings do not include the media (the actual call). The settings on the Media tab should be set accordingly.

Note that if you choose to use a secure connection, both units must support it. Otherwise they cannot negotiate a connection. If an incoming call demands a secure TLS or SIPS connection, the Avaya B179 uses the appropriate protocol even if you have set the phone to use UDP.

Protocol	UDP (User Datagram Protocol) is a protocol on the transport layer in the Internet Protocol Suite. It is a stateless protocol for short messages – datagrams. Stateless implies that it does not establish any connection between sender and receiver in advance. UDP does not guarantee reliability or ordering in the way that TCP does. Datagrams may arrive out of order or go missing without notice. The advantages it offers are speed and efficiency. UDP is the default protocol for SIP. TCP (Transmission Control Protocol) is a protocol on the transport layer in the Internet Protocol Suite. TCP is the standard protocol for

Internet communication. TCP keeps track of all individual packets of data, ensuring that they reach the receiver and are put together properly. TCP is not the default protocol for SIP, because it is slower and uses more bandwidth than UDP.

With UDP and TCP, SIP packets travel in plain text. TLS (Transport Layer Security) is a cryptographic protocol that provides security and data integrity for communications over TCP/IP networks. TLS encrypts the datagrams of the transport layer protocol in use. The secure connection may be to the end device or to the first server (usually the SIP server where the phone is registered). There is no guarantee that there is a secure channel to the end point, but because the SIP server is the only part receiving the user authentication, this is still a rather secure solution.

SIPS (Secure SIP) is a security measure that uses TLS to provide an encrypted end-to-end channel for the SIP messages. To use SIPS, however, both VoIP devices and the SIP server must support it.

ⓘ Even if Transport is set to TLS or SIPS, the Avaya B179 still accepts incoming UDP or TCP signalling.

On phone: ⚙ > SETTINGS > ADVANCED > (PIN) > ACCOUNTS > TRANSPORT (6,2,1,3).

TLS Settings

If you select TLS or SIPS under the transport setting, an additional setting appears on the page. The settings are described below.

It may be possible to use secure communication without a certificate and make changes to these settings. In some cases, if you choose TLS or SIPS, the SIP server requires a certificate for user/client verification. This should be specified in the account information.

You can further increase security by requiring verification of the server, or the client when the Avaya B179 acts as a server for incoming calls. For more information, see Appendix C: Using EAP TLS authentication.

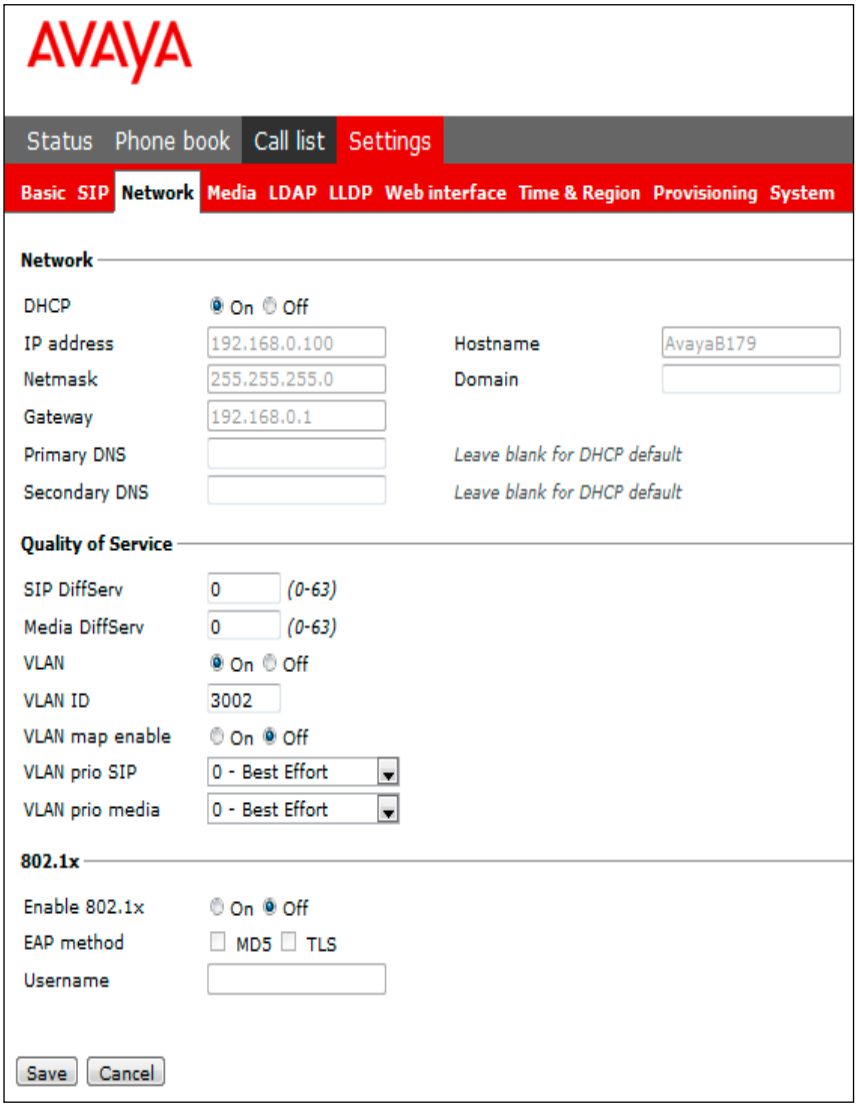
ⓘ The supported encryptions are SHA-1 and SHA-256 1024 and 2048 bits.

Method	The TLS includes a variety of security measures. The methods are defined in the versions of the standard (SSL, SSL v2, SSL v3, TLS v1, TLS v2). The default method is SSLv23, which accepts both SSL v2 and v3.
Negotiation timeout	The TLS settings are negotiated during a call setup (both incoming and outgoing). If this negotiation does not succeed within the specified time (seconds) the negotiation is aborted. Timeout is disabled with 0 (zero).
Verify client	When set to On, the Avaya B179 will activate peer verification for incoming secure SIP connections (TLS or SIPS).

Require client certificate	When set to On, the Avaya B179 rejects incoming secure SIP connections (TLS or SIPS) if the client does not have a valid certificate.
Verify server	When the Avaya B179 is acting as a client (outgoing connections) using secure SIP (TLS or SIPS) it will always receive a certificate from the peer. If Verify server is set to On, the Avaya B179 closes the connection if the server certificate is not valid.
Certificate	Here you can upload a certificate to the Avaya B179 to be used for TLS or SIPS communication. A certificate is a file that combines a public key with information about the owner of the public key, all signed by a trusted third party. If you trust the third party, then you can be sure that the public key belongs to the person/organization named in that file. You can also be sure that everything you decrypt with that public key is encrypted by the person/organization named in the certificate.
Root certificate	The public key in the root certificate is used to verify other certificates. A root certificate is only needed if you have selected client or server verification. A root certificate is signed by the same public key that is in the certificate, a so-called "self-signed" certificate. A typical root certificate is one received from a Certificate Authority.
Private key	Here you can upload a private key to the Avaya B179 to be used for TLS or SIPS communication. A private key is one of the keys in a key-pair used in asymmetric cryptography. Messages encrypted using the public key can only be decrypted using the private key.
Private key password	Password used for encryption of the private key, if it is encrypted.

NETWORK

⇨ Select Settings > Network.



DHCP	Dynamic Host Configuration Protocol is used by network devices (clients) to obtain the parameters necessary for operation in the IP network. This protocol reduces system administration workload, allowing devices to be added to the network with little or no manual configuration. DHCP should be set to On if no other information is given. When set to On, all information on this page will be set automatically.
IP address	IP address of the device (Avaya B179). The address is provided by the network administrator or service provider if DHCP is not in use.
Hostname	Set to avaya as default. Can be changed to suitable name.
Netmask	Usually set to 255.255.255.0 to limit network traffic to the subnet.
Domain	The domain where the device is located. May be left blank.
Gateway	The device or server used for Internet communication.
Primary DNS	The address to the primary DNS (Domain Name System) server - a program or computer that maps a human-recognisable name to its computer-recognisable identifier (IP address).
Secondary DNS	The address of an optional secondary DNS server.

On phone: ⚙ > SETTINGS > ADVANCED > (PIN) > NETWORK (6,2,2).

Quality of Service

Quality of service is used in IP networks to provide different priority to different applicaions,or to guarantee a certain level of performance to a critical data flow such as voice or video. Differentiated Services or DiffServ is a networking architecture that specifies a simple mechanism for classifying network traffic using a 6-bit field in the header of the IP packets. VLAN (Virtual LAN) is a technology to logically divide a physical network into several logical nets and thus to differentiate traffic.

SIP DiffServ	Enter a value between 0 and 63 to prioritize the SIP messages.
Media DiffServ	Enter a value between 0 and 63 to prioritize the media packets (voice).
VLAN	By enabling this option, all communication to and from Avaya B179 is done via the VLAN specified under VLAN ID. Note that this VLAN also must be used to communicate with Avaya B179 via the web interface.
VLAN ID	The ID number to be used for the IP telephony VLAN.
VLAN map enable	Enabling VLAN priority mapping from the DiffServ setting.
VLAN prio SIP	Set a value between 0 and 7 to prioritize the SIP messages in the VLAN.
VLAN prio media	Set a value between 0 and 7 to prioritize the media packets in the VLAN.

On phone: ⚙ > SETTINGS > ADVANCED > (PIN) > NETWORK > ETHERNET > VLAN (6,2,2,2,1).

802.1x

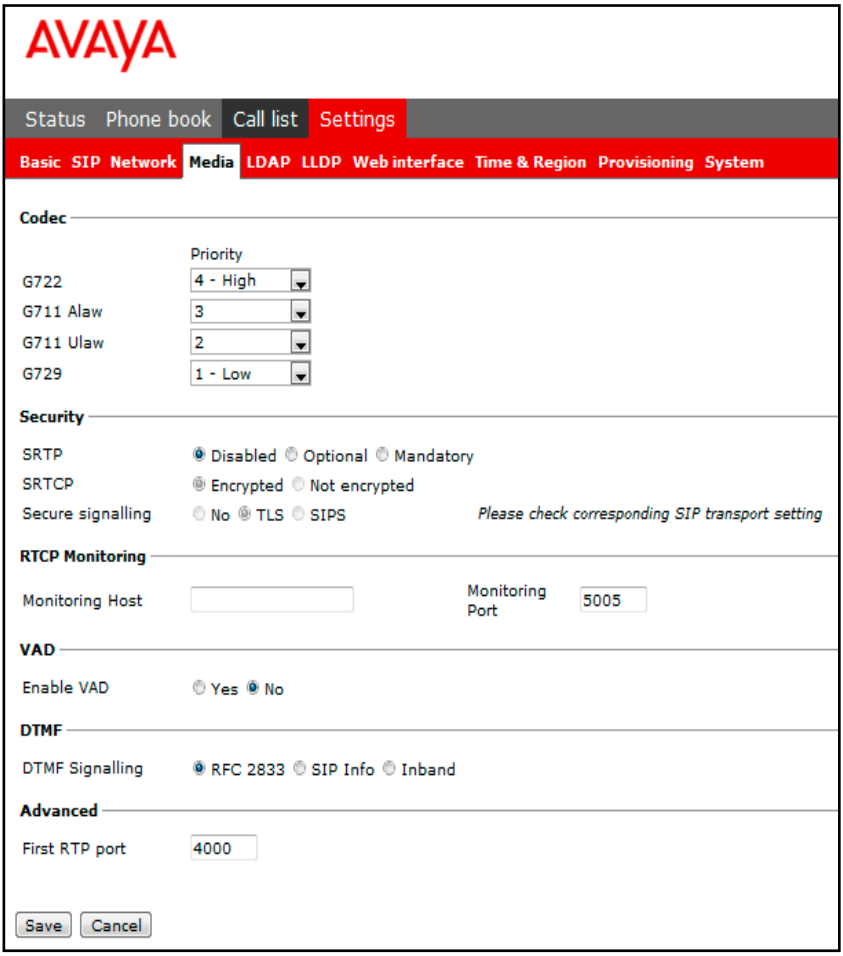
IEEE 802.1X is an IEEE Standard for port-based Network Access Control and is part of the IEEE 802.1 group of networking protocols. It provides an authentication mechanism to devices wishing to attach to a LAN or WLAN.

Enable 802.1x	By enabling this option, Avaya B179 asks an authentication server for permission when connected to the LAN.
EAP method	Select which EAP (Extensible Authentication Protocol) method to use: MD5 or TLS.
Username	The device identity in the network.
MD5 password	Password for the device identity when using MD5.
Certificate	Here you can upload a certificate to the Avaya B179 to be used for authentication when using TLS.
Root certificate	The public key in the root certificate is used to verify other certificates when using TLS.
Private key	Here you can upload a private key to the Avaya B179 to be used for authentication when using TLS.
TLS password	The password used for encryption of the private key when using TLS. On phone: ⚙ > SETTINGS > ADVANCED > (PIN) > NETWORK > ETHERNET > 802.1X AUTH (6,2,2,2,2).

MEDIA

⇨ Select Settings > Media.

The media settings determine how audio is sent between the devices. The devices negotiate via SIP before a call is connected. All devices must support the same media types, codecs and security settings.



Codec

Codecs are used to convert an analog voice signal to a digitally encoded version and vice versa. Codecs vary in the sound quality they deliver and the bandwidth required. The Avaya B179 supports the most common codecs and each codec can be given a precedence depending on your requirements for high quality audio or low bandwidth use.

The priority can be set to from 4 (high) to 1 (low) or 0 (disabled)

G722	G.722 is an ITU-T standard codec that provides 7 kHz wideband audio at a data rate within 64 kbit/s. It offers greately improved speech quality compared with older narrowband codecs such as G.711, but requires a high quality network connection between the devices.
G711 Alaw	G.711 is an ITU-T standard codec that uses audio companding. Companding algorithms reduce the dynamic range of an audio signal. In analog systems, this can increase the signal-to-noise ratio achieved during transmission and, in the digital domain, can reduce the quantization error. Two main compression algorithms are defined in the standard, the μ-law algorithm (used in North America and Japan) and A-law algorithm (used in Europe and the rest of the world).
G711 Ulaw	See G711 A-law above.
G729	G.729 is an ITU-T standard codec that operates at 8 kbit/s. It is mostly used in VoIP applications with low bandwidth requirement.

On phone: ⚙ > SETTINGS > ADVANCED > (PIN) > MEDIA > CODEC (6,2,4,1).

Security

The media in VoIP calls is usually sent using the RTP protocol (Real-time Transport Protocol). RTP is a standardized packet format for delivering audio and video over the Internet.

SRTP (Secure Real-time Transport Protocol) is an extension of RTP to provide encryption, message authentication and integrity for the audio and video streams.

All devices must support SRTP to establish a connection. It is therefore possible to set SRTP as disabled, optional or mandatory.

RTCP can be used to control the RTP session. When using SRTP, the same security-related features are added to the control protocol. SRTCP (Secure Real Time Control Protocol) can be either encrypted or not encrypted.

SRTP	If set to disabled, the media is sent using RTP. Note that despite this setting, the Avaya B179 will still use a secure channel if the opposite device demands it. If set to optional or mandatory, a padlock will be shown in the bottom right-hand corner of the screen. If the other devices support SRTP, the padlock will be locked. Otherwise, an open padlock will be displayed. If set to mandatory, the call will not be connected if the other devices do not support SRTP.
SRTCP	Is automatically used when SRTP is enabled. Can be set to use encryptation or not.
Secure signalling	The SIP messages (signalling) and the SRTP cipher key are sent on a different channel than the media and are not affected by the RTP/SRTP setting. To ensure a secure connection, the signalling must be secured using TLS or SIPS, see page 20. Note that the SIP transport setting must be set accordingly.

On phone: ⚙ > SETTINGS > ADVANCED > (PIN) > MEDIA > SECURITY (6,2,4,4).

RTCP Monitoring

Monitoring Host	IP Address of the RTCP monitoring server. All RTCP packets are sent to this host for monitoring. If you use SRTP, unencrypted packets are sent to this host. In the configuration file, update the IP address in the <rtcp_monitoring_server_host> tag.
Monitoring Port	The port of the RTCP monitoring server.

VAD

Voice Activity Detection (speech detection) is a technique used in speech processing to detect the presence or absence of human speech in regions of audio. In VoIP applications, VAD is mainly used to avoid unnecessary coding and transmission of silence packets, saving on computation and network bandwidth.

On phone: ⚙ > SETTINGS > ADVANCED > (PIN) > MEDIA > VAD (6,2,4,2).

DTMF

DTMF (Dual-tone multi-frequency) signalling is used for telephone signalling over the line to the phone switch or PBX.

If the device itself generates the tones and they are sent in the voice-frequency band, the method is called Inband. This is not the best method when using VoIP. Low bit rate codecs may corrupt the signalling tones and make it difficult for the switch to identify them.

RFC 2833 is a method of carrying DTMF signals in RTP packets using a separate RTP payload format. With this method a PSTN gateway reproduces the DTMF tones sent from the end device.

With SIP Info the DTMF signals are sent as SIP requests. The SIP switch creates the tones if the call is transferred to the PSTN.

Use RFC 2833 or SIP Info as preferred methods. Switch to inband only if you encounter problems using DTMF signalling with your PBX/SIP switch.

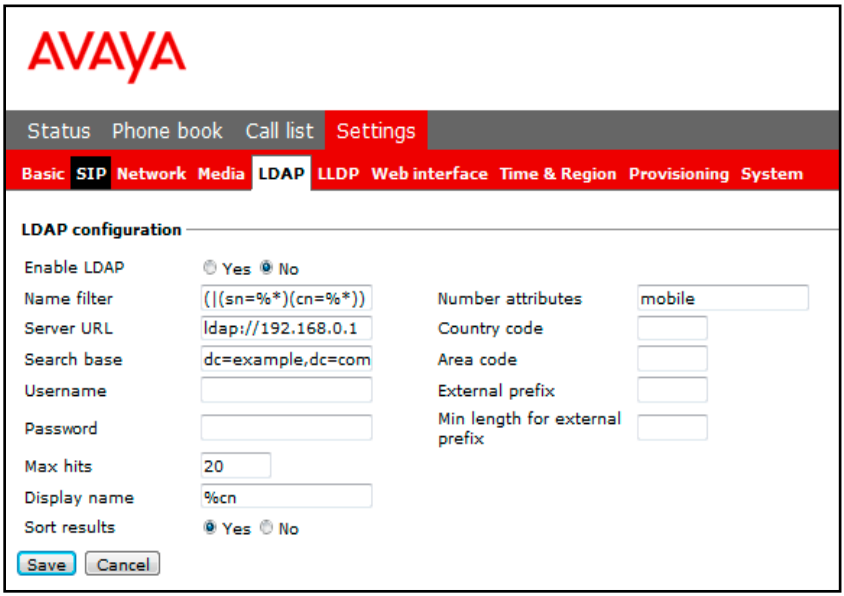
On phone: ⚙ > SETTINGS > ADVANCED > (PIN) > MEDIA > DTMF SIGNALLING (6,2,4,3).

Advanced

First RTP port	If the RTP packets must be directed to a specific port series, the first port number is set here.

LDAP

⇨ Select Settings > LDAP.



Avaya B179 has support for an external phone book, which means it can communicate with a directory server using LDAP (Lightweight Directory Access Protocol). The built in search function dynamically filters the content from the LDAP database, based on the search characters the user enter.

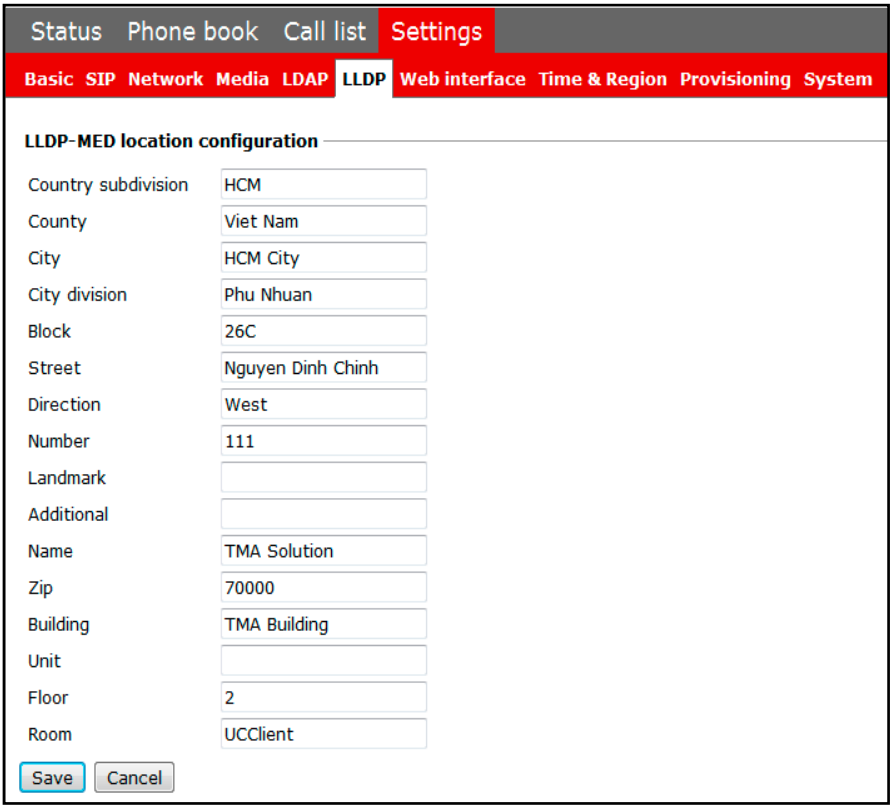
To make the LDAP phone book available, the administrator has to activate and configure the LDAP feature.

Enable LDAP	The LDAP feature is disabled by default because it has to be configured.
Name filter	Defines how the entered search characters are used. The filter is designed conforming to the string representation of LDAP search filters described in RFC2254. The character % in the filter string will be replaced with the search character entered by the user. Example: (|(sn=%*)(cn=%*)) - All entries with the search characters in the beginning of the sn OR cn attribute are presented to the user.
Server URL	The IP address of the LDAP server host. Supports ldap and ldaps.
Search base	The DN (distinguished name) of the search base Example: dc=domain, dc=com.

Username	Leave this field blank if the LDAP server does not require a username.
Password	Leave this field blank if the LDAP server does not require username and password.
Max hits	The maximum number of hits to return for each LDAP search.
Display name	Specifies how the search hits shall be presented on the display in Avaya B179. Example: %cn - shows the cn attribute. %givenName %sn - shows the givenName attribute and the sn attribute with a space in between.
Sort results	Sorts the search hits based on the Display name.
Number attributes	Here you define the attributes that shall be displayed for a selected search hit. Example: mobile telephoneNumber - shows the mobile phone number and office phone number on separate rows for the selected Display name. (Refer to the LDAP administrator for the actual names of the fields in the LDAP database.)
Country code	By entering the country code where the phone is located, the country code in any phone number attribute is ignored, if it is identical.
Area code	By entering the area code where the phone is located, the area code in any phone number attribute is ignored, if it is identical.
External prefix	If a special prefix is needed to dial external numbers, it should be added here. Use this if you for example need to dial 0 to get a dialing tone.
Min length for external prefix	Restricts the external prefix to be added only if the phone number is longer than the min length. This makes it possible to use short internal numbers.
Exact length for no external prefix	The external prefix in not added if the phone number has exactly the entered length.
Number prefix for no external prefix	All numbers that starts with this number will not have the external prefix added. Useful if you know that all internal numbers start with a certain number.

LLDP

⇨ Select Settings > LLDP



Link Layer Discovery Protocol (LLDP) is a data link layer protocol. LLDP defines a standard for Ethernet network devices to broadcast and receive information about other devices in the same network. The information is sent as LLDP Data Units (LLDPDU). Each LLDPDU is a sequence of Time-Length-Value (TLV) strings.

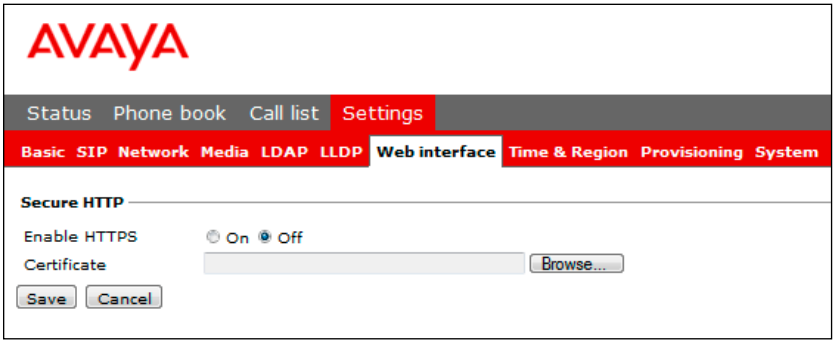
The B179 Conference Phone supports LLDP on primary Ethernet interfaces. The following table lists the TLVs that the B179 Conference Phone supports:

CATEGORY	TLV NAME (TYPE)	STRING LENGTH	TLV INFO STRING (VALUE)
BASIC MANDATORY	CHASSIS ID	6	MAC ADDRESS OF THE PHONE
BASIC MANDATORY	PORT ID	7	IP ADDRESS OF THE PHONE
BASIC MANDATORY	TIME TO LIVE	2	LLDP_TTL
BASIC OPTIONAL	SYSTEM NAME	9	LLDP_SYSTEM_NAME
BASIC OPTIONAL	SYSTEM CAPABILITIES	4	BIT 5 (PHONE) IS SET IN THE SYSTEM CAPABILITIES OCTET. IF THE PHONE IS REGISTERED, BIT 5 IS SET IN THE ENABLED CAPABILITIES OCTET.
BASIC OPTIONAL	MANAGEMENT ADDRESS	23	MGMT ADDR STRING LENGTH = 5; MGMT ADDRESS SUBTYPE = 01; (IPV4) MGMT ADDRESS = IPADD; INTERFACE NUMBER SUBTYPE = 2; INTERFACE NUMBER = 3
ORGANIZATION SPECIFIC IEEE 802.3	MAC/PHY/ CONFIGURATION STATUS	9	802.3 OUI = 00-12-0F (HEX); 802.3 SUBTYPE = 1; AUTONEGOTIATION SUPPORT/ STATUS = VALUE SENT DURING AUTO-NEGOTIATION; OPTIONAL MAU TYPE = LLDP_MAU
TIA LLDP MED	LLDP-MED CAPABILITIES	7	TIA OUI = 00-12-BB (HEX); LLDP CAPABILITIES SUBTYPE = 1; LLDP-MED CAPABILITIES = 00-23 (INVENTORY, NETWORK POLICY, MED CAPS); LLDP-MED DEVICE TYPE = 3 (CLASS III)

CATEGORY	TLV NAME (TYPE)	STRING LENGTH	TLV INFO STRING (VALUE)
TIA LLDP MED	NETWORK POLICY (VOICE)	8	TIA OUI = 00-12-BB (HEX); NETWORK POLICY SUBTYPE = 2; APPLICATION TYPE = 1 (VOICE) U = 0 (NETWORK POLICY IS DEFINED) T = TAGGING X = 0 (RESERVED BIT) VLAN ID = VLAN_IN_USE
TIA LLDP MED	INVENTORY - SOFTWARE REVISION	5 - 36	TIA OUI = 00-12-BB (HEX); SOFTWARE REVISION SUBTYPE = 7; SOFTWARE REVISION = VALUE
TIA LLDP MED	INVENTORY - MANUFACTURER NAME	9	TIA OUI = 00-12-BB (HEX); MANUFACTURER NAME SUBTYPE = 9; MANUFACTURER NAME = VALUE
TIA LLDP MED	INVENTORY - MODEL NAME	8	TIA OUI = 00-12-BB (HEX); MODEL NAME SUBTYPE = 10; MODEL NAME =VALUE
BASIC MANDATORY	END-OF-LLDPU	0	NA

WEB INTERFACE

⇨ Select Settings > Web interface.



The web server in the Avaya B179 supports secure connections using HTTPS.

Enable HTTPS	Set Enable HTTPS to On if you need a secure communication between the PC used for setup and the phone.
Certificate	To use HTTPS you must upload a .PEM certificate to the phone.

ⓘ Note: To convert .DER or .PFX formats to .PEM, use the OpenSSL commands listed in step 6 of Appendix C on page 84.

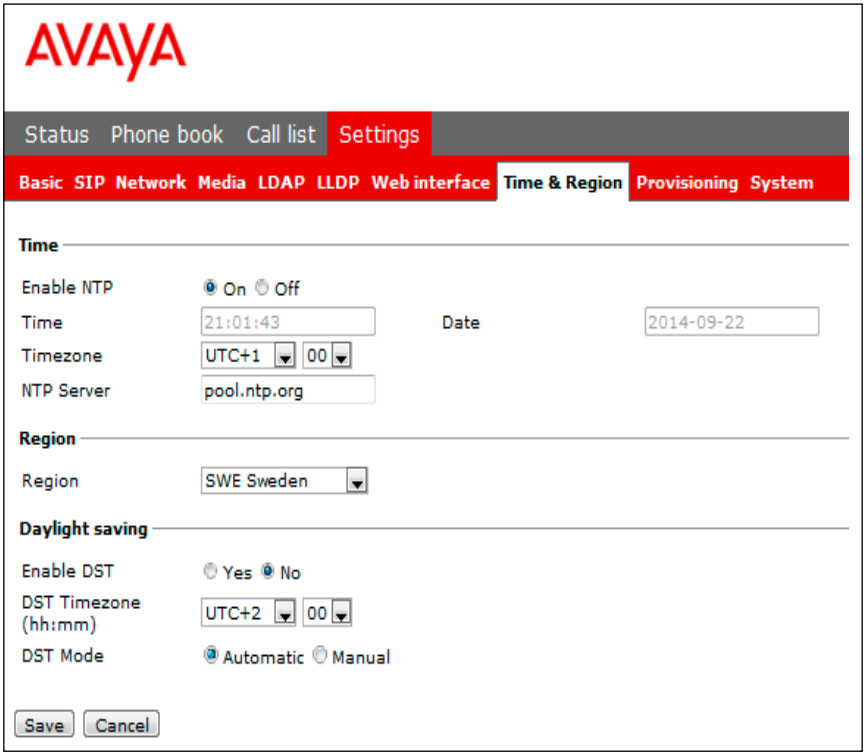
ⓘ Use the following command to generate a HTTPS web interface certificate:

```
openssl req -new -x509 -keyout https_web_certificate.pem -out
https_web_certificate.pem -day<number of days> -nodes
```

On phone: ⚙ > SETTINGS > ADVANCED > (PIN) > WEB INTERFACE (6,2,7).

TIME & REGION

⇨ Select Settings > Time & Region.



Time

Enable NTP	NTP (Network Time Protocol) is a protocol for distributing the Coordinated Universal Time (UTC) by means of synchronizing the clocks of computer systems over packet-switched, variable-latency data networks.
Time	This field shows the actual time if NTP is enabled. Otherwise enter the correct time (hh:mm:ss) and save the setting.
Date	This field shows the actual date if NTP is enabled. Otherwise enter the correct date (yyyy-mm-dd) and save the setting.
Timezone	Select the UTC time zone in your country.

Daylight saving	Select the Yes radio button if DST (Daylight Saving Time or Summer Time) is currently used in your country. Note that this setting only adjusts the time by one hour and does not change the time automatically when the DST starts and ends.
NTP Server	The NTP pool is a dynamic collection of networked computers that volunteer to provide highly accurate time via NTP to clients worldwide. These computers are part of the pool.ntp.org domain and part of several subdomains divided by geographical zones. They are distributed to NTP clients via round robin DNS.

On phone: ✱ > SETTINGS > ADVANCED > (PIN) > TIME (6,2,5).

Region

Select the region where you are. This setting determines the signalling (disconnect tone, busy tone, etc).

On phone: ✱ > SETTINGS > ADVANCED > (PIN) > REGION (6,2,6).

Daylight saving

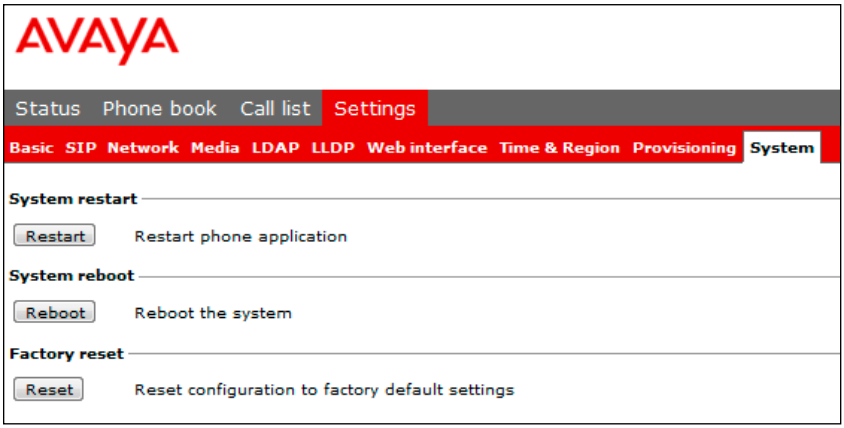
Enable DST	Select the Yes radio button if DST (Daylight Saving Time or Summer Time) is used in your country.
DST Timezone	Select the offset from UTC time when daylight saving is in use.
DST Mode	When set to Automatic, OmniTouch 4135 IP uses dates stored in the phone to adjust for DST. When set to Manual, you need to manually set the offset two times a year.
Start/Stop fixed date	Set to Yes if DST changes the same date every year in your country. Then select the time and date it changes. Set to No if DST changes a specific week and day each eyar. (For instance third sunday in March.) Then select the month, week and time it changes.

PROVISIONING

See “PROVISIONING – UPGRADE AND CONFIGURATION” on page 46.

SYSTEM

⇨ Select Settings > System.



Application restart

The Restart button restarts the phone application. This takes less than 30 seconds.

On phone: ⚙ > SYSTEM > RESTART (7,1).

System reboot

The Reboot button reboots the conference phone. The starting procedure may take about two minutes.

On phone: ⚙ > SYSTEM > REBOOT (7,2).

Factory reset

The Reset button resets the Avaya B179 to factory default settings. All personal settings, including account information, are erased.

On phone: ⚙ > SYSTEM > FACTORY RESET (7,3).

Hard reset to factory settings

See the information about resetting the phone if you have forgotten the Admin PIN code.

CONNECTING A WIRELESS HEADSET

⇨ Connect the headset to the Aux port on Avaya B179.

The microphones from the Avaya B179 and the wireless headset will work simultaneously and transmit the call to other participants in the phone conference.

Please refer to the headset manual for further information.

Turning off the internal speakers when using a headset

The internal speakers can be turned off temporarily if you wish to use the Avaya B179 as a personal telephone with a headset.

⇨ During a call, select ✱ > HEADSET.

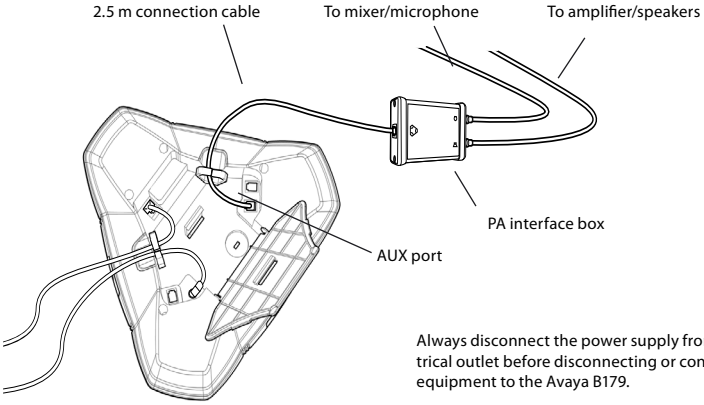
⇨ Select YES when asked “SPEAKER OFF?”.

ⓘ The speakers come on automatically when the call is ended.

HEADSET AND PA INSTALLATION AND SETTINGS

CONNECTING A PA INTERFACE BOX

The Avaya B179 can be connected to an external PA system using a PA interface box.



Always disconnect the power supply from the electrical outlet before disconnecting or connecting equipment to the Avaya B179.

⇨ Connect the PA-box to the AUX port on Avaya B179 with the included cable.

⇨ Connect the external amplifier to the RCA connector marked with a speaker.

⇨ Connect the microphone mixer to the RCA connector marked with a microphone.

Changing the auxilary port setting

⇨ Select Settings > Basic.

⇨ Select PA under the heading Auxilary port to activate the functions for external microphones and speaker system.

⇨ Click on Save.

ⓘ Do not select the PA option unless a PA system is connected. This option turns off the internal microphone and internal speakers as default.

On phone: ✱ > SETTINGS > BASIC > AUX PORT (6,1,5).

PA SETTINGS

To match several types of situations and equipment, there are some settings available in the Avaya B179 menu.

Activating internal microphone and speakers

ⓘ These settings are not available via the web interface.

⇨ Select ✱ > SETTINGS > BASIC > PA (6,1,6).

⇨ Select INTERNAL MIC and press OK to switch between on (shaded box) and off.

ⓘ To ensure maximum audio quality, do not use the internal microphone and external microphones connected via the PA box at the same time.

ⓘ Only the internal microphone is turned off. Any external microphones connected to the Avaya B179 are still turned on.

⇨ Select INTERNAL SPKR and press OK to switch between on (shaded box) and off.

ⓘ To ensure maximum audio quality, do not use the internal speakers and external speakers connected via the PA box at the same time.

Adjusting microphone volume from PA

⇨ During a call, select ✱ > PA > PA MONITOR.

⇨ Adjust the microphone volume from the mixer so that the level on the display screen is around 10–12 when speaking in a normal tone.

Adjusting PA calibration manually

It is possible to calibrate the duplex performance of the conference phone when it is connected to a PA system. The calibration level can be set automatically by the Avaya B179 or adjusted manually to any value between 0 and 5 (0 being full duplex).

- Increase the calibration if the other party experiences disturbing echo.
- Decrease the calibration if the other party experiences low duplex, i.e. your voice is muted or clipped when the other party is speaking.

ⓘ The position of the PA system's microphones and speakers and the amplifier's settings may affect full duplex performance.

⇨ Select ✱ > PA > CALIBRATION.

ⓘ NB. You must ask the person you are calling to assess the effect of the adjustments you make.

ⓘ AUTO is the default setting. The figure shown in brackets is the measured calibration value.

⇨ Select different levels and compare the audio quality to achieve your preferred setting.

HARD SYSTEM RECOVERY

Reset configuration

If you have forgotten the Admin PIN code, the only way to reset it to default is to do a hard factory reset. This is the same as the Factory reset in the system menu (> SYSTEM > FACTORY RESET).

ⓘ This erases all settings including account information and contacts!

⇨ Disconnect the power supply cable. Note that this is the same as the network cable if the phone uses Power over Ethernet.

⇨ Press and hold the button while you connect the cable again (Restarts the Avaya B179). Hold the button pressed until the SYSTEM RECOVERY menu is shown on the display.

ⓘ You can press any other button than 1, 2, or 3 to start the phone without resetting.

⇨ Press 1 to select Reset configuration and confirm with OK.

⇨ Upgrade to the latest version of the software when the phone has started and redo the setup of account and other settings.

Restore firmware

This replaces the current software with the one supplied with the phone. All settings are erased.

⇨ Export a configuration file if local settings should be saved and export the contacts if these are stored in the unit.

⇨ Disconnect the power supply cable. Note that this is the same as the network cable if the phone uses Power over Ethernet.

⇨ Press and hold the button while you connect the cable again (restarts the Avaya B179). Hold the button pressed until the SYSTEM RECOVERY menu is shown on the display.

ⓘ You can press any other button than 1, 2, or 3 to start the phone without restoring the firmware.

⇨ Press 3 to select Restore firmware and confirm with OK.

All content in the phone's memory is erased and the firmware supplied with the phone is written to the memory.

⇨ Upgrade to the preferred version of the firmware when the phone has started.

⇨ Import the local configuration and previous contacts or do a manual account setup.

PROVISIONING – UPGRADE AND CONFIGURATION

FIRMWARE UPGRADE ON A SINGLE PHONE

The easiest way to upgrade the Avaya B179 is via a computer connected to the same network. Via the web interface, you can check for a more recent version and then automatically install it.

It is also possible to download the latest version, via the Avaya website (support.avaya.com), and then install the file via the web interface or using a SD card.

Using the web interface

⇨ Select Settings > Provisioning.

Upgrading from downloaded file

It is possible to download a firmware file from support.avaya.com and install it on the Avaya B179 from the local hard disk.

⇨ Download the firmware file from support.avaya.com.

⇨ Click on the Browse… button and locate and select the downloaded file.

⇨ Click on the Upgrade button.

Upgrading from SD card

Upgrading from SD card may be suitable if you have many phones to upgrade. The phones do not have to be connected to the network.

⇨ Download the latest firmware as above and save it on a SD card.

⇨ Put the SD card in the phone you want to upgrade.

⇨ Disconnect the power supply cable. Note that this is the same as the network cable if the phone uses Power over Ethernet.

⇨ Press and hold the ✱ button while you connect the cable again (i.e. starts the Avaya B179). Hold the button pressed until the SYSTEM RECOVERY menu is shown on the display.

ⓘ You can press any other button than 1, 2, or 3 to start the phone without any change.

⇨ Press 2 to select SD-card upgrade.

The Avaya B179 is upgraded with the firmware file on the SD card and starts when the upgrade is done.

After upgrading

If DHCP is used in the network, the IP address may have been changed. If the web browser loses contact with Avaya B179, check the IP address on the conference phone, see USING THE WEB INTERFACE.

DISABLING FIRMWARE UPGRADE FROM SD CARD

To disable the firmware upgrade from the SD card, set the <sd_upgrade_disabled> tag in the global configuration file to true.

Note: You cannot enable the firmware upgrade from the SD card after it is disabled.

The recovery process prompts to initiate the SD card-based firmware upgrade even if the option is disabled in the configuration file. When the user selects the option to initiate the upgrade, the following messages are displayed:

SD-card upgrade. Please wait.

SD-Upgrade disabled. Skipping upgrade.

FIRMWARE UPGRADE ON MULTIPLE PHONES

To upgrade the firmware on multiple B179 conference phones, use a device management server.. For information about configuring and using a device management server, see Using a device management server on Page 58.

You can save a configuration xml file to be used as:

- Backup (i.e. if the system has been reset to factory default)
- Configuration interface (there are some settings that are not configurable via the web interface)
- Management tool (export, edit and import settings to a set of phones instead of doing the settings on each phone)
- Use with a Device Management server, see page 48.

The structure of the xml file is as follows:

<locale>	
<region>	
<recording>	
<enable>	
<logging>	
<level>	The phone application logs messages to log facility LOCAL0. Log level 1–5 (equivalent to Fatal–Trace)
<log_sip>	Log SIP messages to log facility LOCAL1. Default is true.
<remote_log>	Log messages to a remote log server. Default is false.
<remote_host />	Remote log server.
<network>	
<net>	
<dhcp>	Specify if DHCP should be used to obtain network settings. If so, the other network settings won't be used.
<ip>	Specify the IP address of the Avaya B179.
<netmask>	The netmask of the IP address.
<gateway>	Specify the default gateway to be used.
<dns1>	Specify at most two Domain Name Servers to be used.
<dns2>	
<hostname>	Specify host name.
<domain />	Specify domain name.
<vlan>	
<enable>	Virtual LAN enabled if set to true
<id>	VLAN ID.
<std_prio_map>	
<sip_priority>	
<media_priority>	
<ether_8021x>	
<enable>	
<username />	
<eap_md5>	

```
                    <enable>
                    <password>
                <eap_tls>
                    <enable>
                    <password>
    <qos>
        <dscp_sip>
        <dscp_media>
    <time>
        <ntp>
        <timezone>
        <daylight_save>
        <ntps>
```

<sip>

<udp_transport>	Specify if UDP shall be used as transport.
<udp_port>	Specify the UDP port to listen to.
<tcp_transport>	Specify if TCP shall be used as transport.
<tcp_port>	Specify the TCP port to listen to.
<tls_transport>	Specify if TLS shall be used as transport.
<sips_transport>	Specify if SIPS shall be used as transport.
<tls_port>	Specify the TLS port to listen to.
<rtp_port>	Specify the start port for RTP traffic.
<rtp_monitoring_server_host>	Specify the server to which the RTCP packets must be sent.
<outbound_proxy />	Specify the URL of outbound proxies to visit for all outgoing requests. The outbound proxies will be used for all accounts and will be used to build the route set for outgoing requests. The final route set for outgoing requests will consist of the outbound proxies and the proxy configured in the account.
<use_stun>	Use Simple Traversal of UDP through NATs (STUN) for NAT traversal. Default is no.
<stun_domain />	Specify domain name to be resolved with DNS SRV resolution to get the address of the STUN servers. Alternatively application may specify stun_host and stun_relay_host instead.
<stun_host />	Specify STUN server to be used in "HOST[:PORT]" format. If port is not specified, default port 3478 will be used.
<use_turn>	Use Traversal Using Relay NAT (TURN) for NAT traversal. Default is no.
<turn_host />	Specify TURN relay server to be used.
<turn_tcp>	Use TCP connection to TURN server. Default is false.
<turn_user />	TURN username.
<turn_passwd />	TURN password.
<nat_type_in_sdp>	Support for adding and parsing NAT type in the SDP to assist troubleshooting. The valid values are: 0: no information will be added in SDP and parsing is disabled 1: only the NAT type number is added

	2: add both NAT type number and name
<require_100rel>	Specify whether support for reliable provisional response (100rel and PRACK) should be required by default. Note that this setting can be further customized in account configuration.
<use_srtp>	Specify default value of secure media transport usage. Note that this setting can be further customized in account configuration. 0: SRTP will be disabled, and the transport will reject RTP/SAVP offer. 1: SRTP will be advertised as optional and incoming SRTP offer will be accepted. 2: The transport will require that RTP/SAVP media shall be used.
<srtp_secure_signaling>	Specify whether SRTP requires secure signalling. This option is only used when use_srtp option above is non-zero. Note that this setting can be further customized in account configuration. 0: SRTP does not require secure signalling 1: SRTP requires secure transport such as TLS 2: SRTP requires secure end-to-end transport (SIPS)
<codec>	
<type>	Codec type
<name>	Codec name
<prio>	Codec priority (0-4)
<dtmf>	DTMF signalling. Default is 2. 0: In-band 1: SIP message 2: RTP message
<no_vad>	Disable VAD. Default is VAD enabled.
<ec_tail>	Echo canceller tail length, in milliseconds.
<enable_ice>	Enable ICE?
<enable_relay>	Enable ICE relay?
<enable_presence>	Enable the use of presence signalling.
<enable_sip_replaces>	
<enable_blind_transfer>	
<allow_contact_rewrite>	
<tls>	
<tls_password />	Password for the private key
<tls_method>	TLS protocol method from pjsip_ssl_method, which can be: 0: Default (SSLv23) 1: TLSv1 2: SSLv2 3: SSLv3 23: SSLv23
<tls_verify_server>	Verify server certificate.
<tls_verify_client>	Verify client certificate.
<tls_require_client_cert>	Require client certificate.
<tls_neg_timeout>	TLS negotiation timeout in seconds to be applied for both outgoing and incoming connections. If zero, no timeout is used.

<account1>	
<valid>	If this account information is valid or not.
<name>	User defined name of the account
<id>	The full SIP URL for the account.
<registrar>	This is the URL to be put in the request URI for the registration.
<publish_enabled>	If this flag is set, the presence information of this account will be published to the server where the account belongs.
<initial_auth>	If this flag is set, the authentication client framework will send an empty Authorization header in each initial request.
<initial_algo />	Specify the algorithm to use when empty Authorization header is to be sent for each initial request (see above).
<pidf_tuple_id />	Optional PIDF tuple ID for outgoing PUBLISH and NOTIFY. If this value is not specified, a random string will be used.
<force_contact />	Optional URI to be put as Contact for this account. Leave this field empty, so that the value will be calculated automatically based on the transport address.
<require_100rel>	Specify whether support for reliable provisional response (100rel and PRACK) should be required for all sessions of this account.
<proxy_uri />	Optional URI of the proxies to be visited for all outgoing requests that are using this account (REGISTER, INVITE, etc).
<reg_timeout>	Optional interval for registration, in seconds. If the value is zero, default interval will be used.
<cred>	Array of credentials. Normally, if registration is required, at least one credential should be specified to successfully authenticate the service provider. More credentials can be specified, for example when it is expected that requests will be challenged by the proxies in the route set.
<realm>	Realm. Use "*" to make a credential that can be used to authenticate any challenges.
<scheme />	Scheme (e.g. "digest").
<username>	Authentication name
<cred_data_type>	Type of data (0 for plaintext password).
<cred_data>	The data, which can be a plaintext password or a hashed digest.
<auto_update_nat>	This option is useful for keeping the UDP transport address up to date with the NAT public mapped address. When this option is enabled and STUN is configured, the library will keep track of the public IP address from the response of REGISTER request. Once it detects that the address has changed, it will unregister current Contact, update the UDP transport address and register a new Contact to the registrar.
<ka_interval>	Set the interval for periodic keep-alive transmission for this account. If this value is zero, keep-alive will be disabled for this account. The keep-alive transmission will be sent to the registrar's address after successful registration.
<ka_data />	Specify the data to be transmitted as keep-alive packets. Default: CR-LF.
<use_srtp>	Specify whether secure media transport should be used for this account. 0: SRTP will be disabled and the transport will reject RTP/SAVP offer. 1: SRTP will be advertised as optional and incoming SRTP offer will be accepted. 2: The transport will require that RTP/SAVP media is used.
<use_srtcp>	Specifies whether encryption or no encryption is used for the control protocol.

	Valid values: true or false.
<srtp_secure_signaling>	Specify whether SRTP requires secure signalling. This option is only used when use_srtp option above is non-zero. 0: SRTP does not require secure signalling 1: SRTP requires secure transport such as TLS 2: SRTP requires secure end-to-end transport (SIPS)

<account2>

Same as above for account 2

<provisioning>

<upgrade>	
<url>	Place to find software upgrades. The supported URL types are: HTTP, FTP, and TFTP.
<dev_mgnt>	
<enable>	Device management enabled, true or false.
<use_dhcp_option>	Use DHCP option for DM server address.
<dhcp_option>	Specification of which DHCP option to use.
<file_server_address>	DM server address if not provided by DHCP option.
<pagename />	Base name of configuration files to download
<type />	Configuration file type specification
<update_interval>	Timing for downloading files. Shall be entered in crontab format: * * * * * where the * stands for minute (0–59), hour (0–23), day of month (1–30), month (1–12), day of week (0–7) (Sunday =0 or 7) Example: 0 6 * * * = the files are downloaded daily at 6:00.
<https_check_srv_cert>	Controls server certificate, true or false.
<https_protocol>	Possibility to set https protocol if open-ssl auto detection fails.

<www>

<enable_https>	Secure communication to the Avaya B179 web server. Default is false.

<pa>

<enable_pa>	PA enabled, true or false
<enable_internal_mic>	Internal mic enabled when PA set to true.
<enable_internal_spkr>	Internal speakers enabled when PA set to true.
<calibration>	Calibration value. Note that 0 is auto, 1 is calibration value 1, 2 is calibration value 1, etc.

<ldap>

<enable>	LDAP enabled, true or false.
<name_filter>	Name filter according to RFC2254
<server_url>	LDAP server address
<search_base>	The DN (distinguished name) of the search base
<username />	
<password />	
<max_hits>	

<country_code>

<area_code>

<external_prefix />

<min_length_for_ext_prefix />

<exact_length_for_no_ext_prefix />

<number_prefix_for_no_ext_prefix />

<number_attributes>

<display_name>

<sort_results>

<lldp>
<latitude></latitude>
<longitude></longitude>
<altitude></altitude>
<datum></datum>
<language></language>
<country_subdivision></country_subdivision>
<county></county>
<city></city>
<city_division></city_division>
<block></block>
<street></street>
<direction></direction>
<trailing_street_suffix></trailing_street_suffix>
<street_suffix></street_suffix>
<number></number>
<number_suffix></number_suffix>
<landmark></landmark>
<additional></additional>
<name></name>
<zip></zip>
<building></building>
<unit></unit>
<floor></floor>
<room></room>
<place_type></place_type>
<scrip></scrip>
<elin></elin>

Export configuration

⇨ Select Settings > Provisioning.

⇨ Click on the Export button under Configuration.
The configuration file is shown in the web browser.

⇨ Choose to save the page as an xml file.
The xml file is as default saved in your folder for downloaded files.

⇨ If necessary, edit the xml file in a suitable editor.

Import configuration

⇨ Click on the Browse button under Configuration.

⇨ Select the XML file and choose to open it.

⇨ Click on the Import button.

USING A DEVICE MANAGEMENT SERVER

Using Device management facilitates the upgrading and configuration of multiple conference phones. To use this feature, the Device management needs to be enabled (default) and configured and the appropriate files must be located on a server reachable from all phones, here called a device management server.

The configuration and firmware download are controlled with a configurable frequency. The default value is once every 30 minutes. (Note: The interval can only be edited directly in the configuration file.)

Configuration priorities

Because the same configuration parameters can be entered in multiple locations, there is a need for priorities. The local configuration files have the highest priority followed by the global configuration file. Configuration entered on the unit itself, via the web interface or directly on the phone, is overridden the next time the configuration files are downloaded.

ⓘ Note one exception. Phone language entered on the unit will take precedence.

Files on the device management server

Global configuration file

The global configuration file contains the basic configuration – all settings that are common for all conference phones on your location. The easiest way to create this file is simply to configure one phone and export the configuration file or use the built in configuration file creator.

The default name for this file is avayab179.xml, but it is possible to create a custom name by using the pagename element in the configuration file. It is also possible to refer to a cgi, php, asp, js or jsp file, instead of the xml file, if this is declared using the type element in the configuration file.

Local configuration file

The local configuration file contains configuration parameters that are unique for every conference phone. The settings in this file takes precedence over the settings in the global configuration file. The default name for this file is avaya-<MAC>.xml, where <MAC> is the MAC address of the specific conference phone. The MAC address should be written without colons.

It is possible to create a custom name by using the pagename element in the configuration file. It is also possible to refer to a cgi, php, asp, js or jsp file, instead of the xml file, if this is declared using the type element in the configuration file.

Avaya B179 searches for configuration files in the following order:

''	Type parameter value	Result
1	<nothing>	<pagename>.xml
2	cgi	<pagename>.cgi?phone_model=avaya_b179>
3	%"%	<pagename>.php?phone_model=avaya_b179>
4	asp	<pagename>.asp?phone_model=avaya_b179>
5	E	<pagename>.js?phone_model=avaya_b179>
6	E %	<pagename>.jsp?phone_model=avaya_b179>
:	<any name>	<pagename>.<any name>? phone_model=avaya_b179
=	auto	1, 2, 3, 4, 5 and 6 will be tried in that order

Firmware binary

Contains the firmware binary that will be downloaded and installed by Avaya B179 if the metadata file shows that this is a newer version than the present installed. The binary file can be downloaded from support.avaya.com.

Firmware metadata file

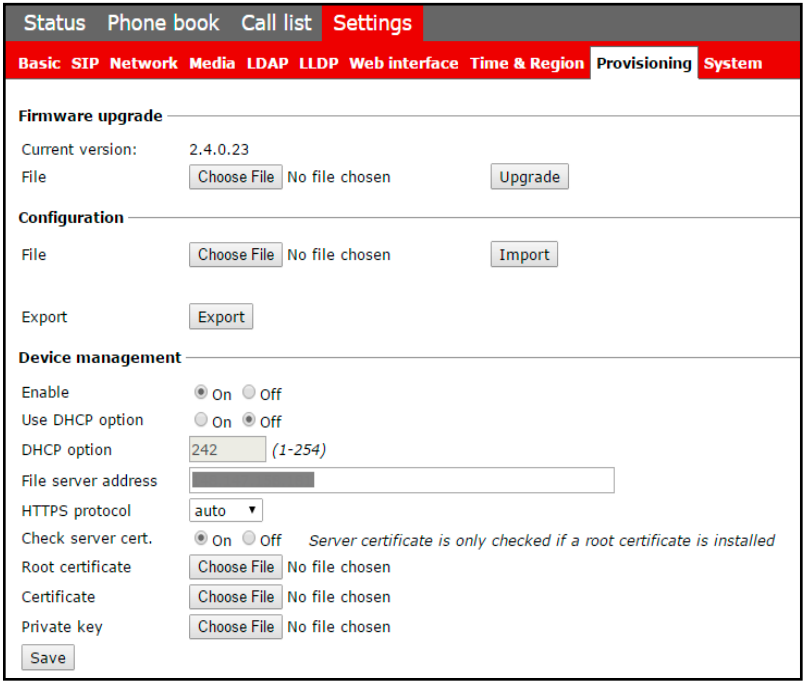
A metadata file in xml format with information of the firmware version in the binary file. The file is used to check if the binary file should be downloaded to the phone or not.

The name of this file shall be avaya_fw_version.xml. The file shall contain the following elements in xml format.

<firmware_version>
<version>X.X.X </version> Eg. 2.3.9
<filename>xxxx </filename> Eg. AVAYA_B179_v2.3.9.kt
<checksum>XXXX </checksum> MD5 checksum of the firmware binary
</firmware_version>

DEVICE MANAGEMENT CONFIGURATION IN AVAYA B179

⇨ Select Settings > Provisioning.



Device management

Enable	On enables Device management.
Use DHCP option	Set to on if you want to use DHCP option for DM server address.
DHCP option	Enter the DHCP option used for the DM server address. 242: Avaya specific option (default) 56: DHCP message 60: Class Id 61: Client Id 66: Server-name 67: Bootfile-name

File server address	DM server address if not provided by DHCP option.
HTTPS protocol	Default is auto, but can be set to SSLv2 or SSLv3 if open-ssl auto detection fails.
Check server cert.	Enable authentication with certificate.
Certificate	Here you can upload a certificate to the Avaya B179 to be used for authentication when using Device management.
Root certificate	The public key in the root certificate is used to verify other certificates when using Device management.
Private key	Here you can upload a private key to the Avaya B179 to be used for authentication when using Device management.

Setting up a Device management server

This is a description of a manual method to create the configuration files.

⇨ Select Settings > Provisioning.

⇨ Enable Device management and enter the server information.

Creating a global configuration file

⇨ Configure one phone with the basic configuration.

⇨ Click on Export to create a configuratuion file.

⇨ If necessary, edit the xml file in a suitable editor.

ⓘ Some parameters can't be entered via the web interface (update frequency, pagename, and filetype).

ⓘ To avoid confusion, it may be wise to delete the local information from the file (eg. account information).

⇨ Save the file with the name avayab179.xml on the File server address specified above.

Creating local configuration files

⇨ Save a copy of the configuration file for each conference phone on your location with content only in the elements that shall be unique for each conference phone (eg. account information).

The default name for each file is avaya-<MAC>.xml, where <MAC> is the MAC address of the specific conference phone.

⇨ Place the configuration files on the File server address specified above.

Firmware binary

⇨ Place the firmware binary file on the Provisioning server.

⇨ Create a Firmware metadata file according to page 49 and place it on the File server address specified above.

ⓘ Depending on the server used, and the security settings, there might be necessary to add the file type .kt to the MIME settings on the server. This is easily checked by trying to download the kt file from a web browser.

FALL BACK SERVER SUPPORT

Avaya B179 registers concurrently with the primary and secondary proxy servers. The phone also supports provisioning of a third-party fall back server when a connection with the primary or secondary server cannot be established. You can configure the third-party server details by using the web interface and the configuration file.

UPGRADING IN IP OFFICE

Avaya B179 supports the IP Office check-sync message with file ID 4 for firmware upgrades. Please see the IP Office documentation for more information.

HOW TO DO A DOWNGRADE

The only way to “downgrade” – install a previous version of the firmware – is to restore the firmware to factory default and then install the preferred firmware.

IMPORTING AND EXPORTING CONTACTS

You can import contacts from a comma separated values (CSV) file.

ⓘ The maximum number of contacts allowed in the B179 phone book is 1000. During importing, the existing contacts are retained.

Importing contacts

⇨ Select Phone Book.

⇨ Click the Scroll button under the heading Import in the web window.

⇨ Open your CSV file.

⇨ Click on Import.

ⓘ The name is limited to 15 characters, since the Avaya B179 screen cannot display more than 15 characters.

You can export contact books stored in your PC in CSV format.

ⓘ The way the number can be written may depend on the SIP PBX being used, but you can use:
Complete phone number, including country code
Phone number, including area code
Local phone number only
Internal speed dial number (with company's own PBX)
URI, e.g. sip:user@company.com
URI with IP address, e.g. sip:10.10.1.100 (within a local network)

Exporting contacts

You can export your contacts as a CSV document in order to import them into another phone.

⇨ Click on Export.

⇨ Save the document.

IMPORTING AND EXPORTING CONFERENCE GROUPS

The conference group feature requires that your system allows multiple call appearances.

Avaya Aura Communication Manager supports this while Avaya IP Office and Communication Server 1000 do not.

The conference groups can be imported and exported in the same way as the contacts in the phone book, but use a three column csv instead of a two column csv.

ⓘ The maximum number of conference groups allowed in the B179 phone book is 20. During importing, the existing conference groups are retained.

ⓘ The Conference group feature requires that your system allows multiple call appearances. Avaya Aura Communication Manager supports this while Avaya IP Office and Communication Server 1000 do not.

The conference groups can be imported and exported in the same way as the contacts in the phone book, but use a three column csv instead of a two column csv.

You can import and export up to 20 conf groups

COMMUNICATION SERVER 1000 BASED CONFERENCE

You can configure the B179 Conference Phone to use a Avaya Communication Server 1000 (CS1000) for conference calls. A user can press the Conference button on the phone to invoke the CS1000 server for the conference call.

To use the CS1000 server for conference calls:

⇨ Open the configuration file in edit mode.

⇨ In the <sip> tag, add the server IP address in the <conference_server> tag.

In the web interface, go to Settings > SIP.

⇨ Under Advanced, type the address of the server in the Conference Server field.

ⓘ For information about configuring CS1000 server, see Appendix B: Configuring CS1000 Server for B179.

TECHNICAL DATA

Size	Diameter 240 mm, height 77 mm
Weight	1 kg
Color	Liquorice black
Display screen	Illuminated graphics (LCD), 128x64
Keypad	Alphanumerical 0–9, *, on, off, mute, hold, volume up, volume down, 5 buttons for menu navigation, line mode, conference guide
Anti-theft protection	Kensington security slot
Memory	Support for SD and SDHC memory cards (Recording capacity 35h/GB)

Connectivity

Network connection	Modular 8P8C (RJ45), Ethernet 10/100 Base T
Power supply	AC adapter 100–240 V AC/14 V DC IEEE 802.3af Power over Ethernet, Class III.
Extra microphones	2 modular 4P4C
Auxiliary	Modular 4P4C for wireless headset

Network and communication

Network addressing	DHCP and static IP
NAT traversal	STUN, ICE and TURN
Connection protocol	SIP 2.0 (RFC 3261 and companion RFCs)
Transport	UDP, TCP, TLS and SIPS
Security	SRTP and TLS
Quality of Service	DiffServ, VLAN and 802.1x
Audio support	Codecs: G722, G711 A-law, G711 µ-law, G729ab
DTMF tone generation	RFC, SIP INFO, In-band
Time servers	NTP and SNTP
Daylight saving: Configurable for automatic adjustments.	
Configuration	Via integrated web server

Configuration and provisioning

Configuration	Via integrated web server, HTTP or HTTPS
	Separate user and administrator login for secure configuration.
Device management	Support for device management for easy configuration and updating of multiple conference phones.

Directory

Internal phone book	1,000 entries per profile (4 password protected profiles) Export/import of directory Call list
External directory	Support for LDAP

Sound

Technology	OmniSound® Wideband
Microphone	Omni-directional
Reception area	Up to 30 metres^2, >10 people
Speaker	Frequency band 200–7000 Hz,
Volume	90 dB SPL 0.5 m
Equalizer	Three pitches: soft, neutral, bright

Environment

Temperature:	5°–40°C
Relative humidity:	20-80% condensation free
Recommended acoustic conditions:	
	Reverberation period: 0.5 S Rt 60
	Background noise: 45 dBA

Approvals

Electrical safety	EN 60950-1:2006, ANSI/UL 60950-1-2002, CAN/CSA-C22.2, No. 60950-1-03
EMC/Radio	EN 301 489-3 V1.4.1 (2002-08), EN 301 489-1 V1.6.1 (2005-09), FCC Part 15 subpart B class A, FCC Part 15 subpart C, EN 300220-1:2000, EN 300220-2:2000 RoHS

APPENDIX A: REGISTERING B179 CONFERENCE PHONES

To register a B179 Conference Phone to the Avaya network, you must create a communication profile for the phone. The communication profile contains a Avaya Aura Communication Manager endpoint profile and a Avaya Aura Session Manager profile.

The Communication Manager endpoint profile associates the user with a station on a Communication Manager.

The Session Manager profile creates a unique user identity and assigns the primary and the secondary Session Manager, relevant application sequences, and the survivability server.

CONFIGURING THE SESSION MANAGER PROFILE

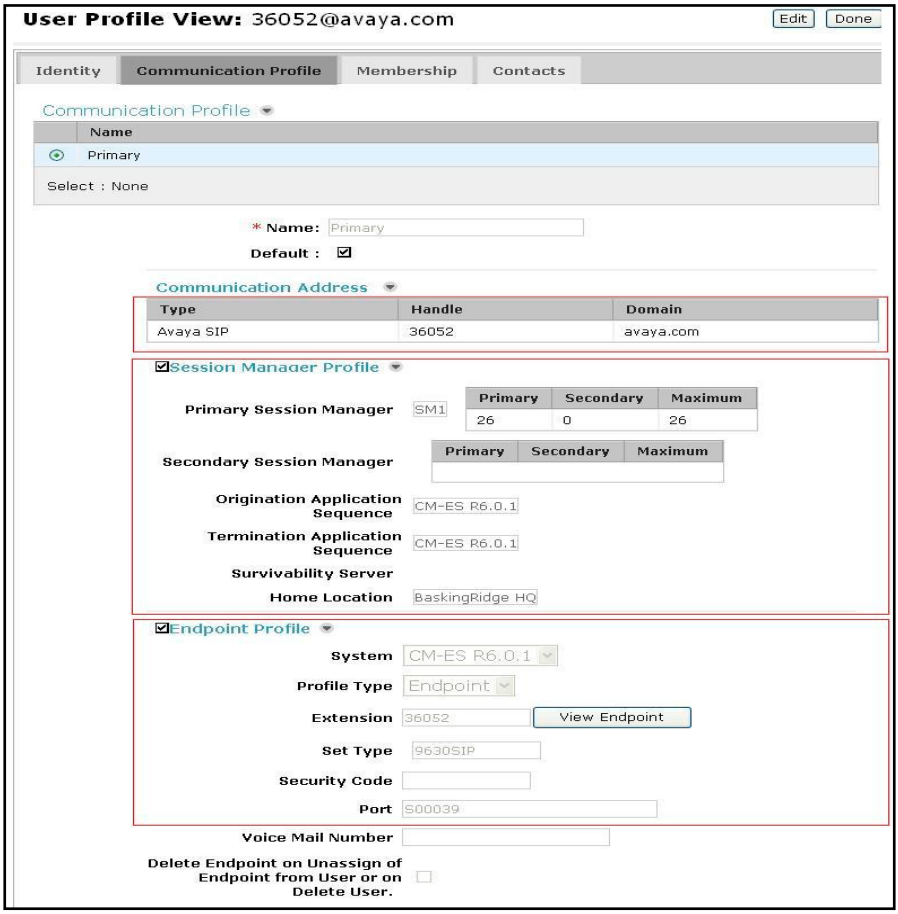
Procedure

On the System Manager web console, under Users, click User Management > Manage Users. A list of users is displayed.

On the User Management page, Click New to create a new endpoint.

Configure the Identity tab.

Configure the Communication Profile tab.



When you add a user in Session Manager, System Manager automatically creates a station in Communication Manager.

CONFIGURING THE COMMUNICATION MANAGER PROFILE

About this task

To use the conference feature of the B179 Conference Phone, the phone must have atleast 4 call appearances. To assign the call appearances to the B179 phone, configure the station associated with the Communication Manager endpoint profile of the phone.

When you add a user in Session Manager, System Manager automatically creates a station in Communication Manager. Use the System Administration Terminal (SAT) interface in Communication Manager to assign 4 call appearances to each B179 station.

Before you begin

Obtain the user credentials for Communication Manager.

Obtain the station ID of the B179 phone.

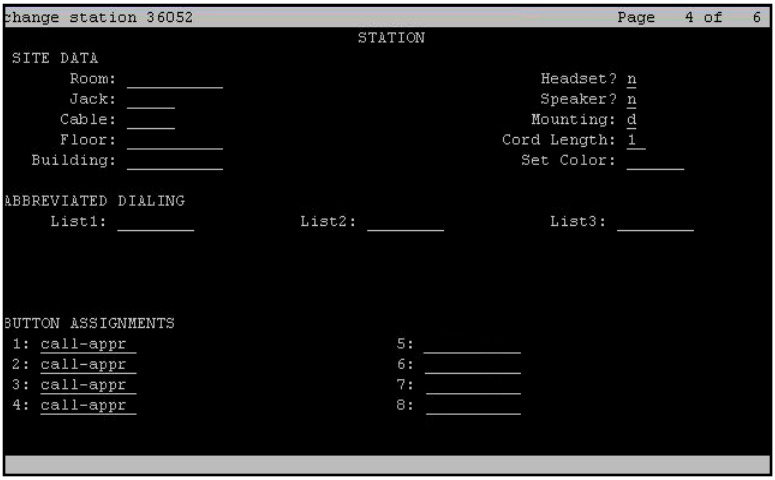
Procedure

Log in to the Communication Manager.

To open the SAT interface, type sat.

At the command: prompt, type change station <station ID>.

Under BUTTON ASSIGNMENTS, assign 4 call appearnces to the phone.



Upgrade log

Logs the upgrade procedure.

Next Steps

On the phone, check that the square icon next to the extension number is solid.

An empty square indicates that the phone is not registered.

APPENDIX B: CONFIGURING CS1000 SERVER FOR B179

Before you begin

Install and configure the CS1000 call server, signaling server, and nodes.

Obtain the administration credentials of the CS1000 call server.

About this task

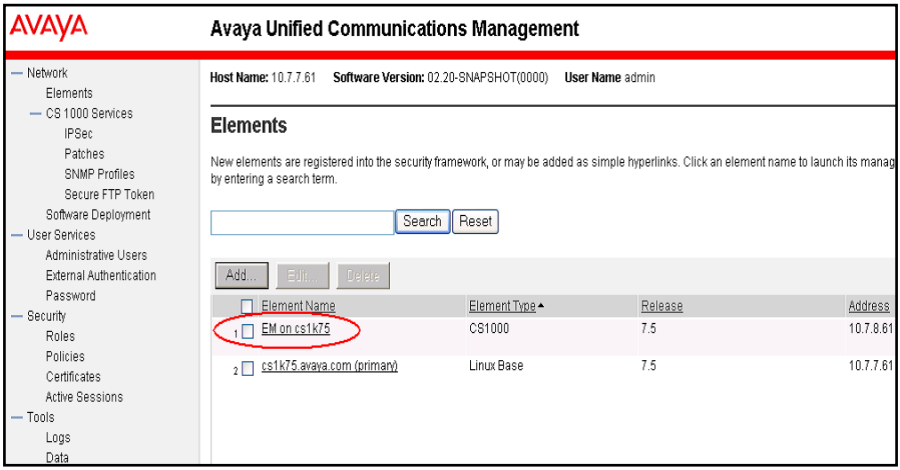
Using CS1000 Element Manager, configure the following components of the CS1000 Server:

- SIP Line service
- SIP Line D-Channel
- Application Module Link (AML)
- Value Added Server (VAS)
- Zone for SIP phones
- SIP Line Route Data Block (RDB)
- SIP Line Virtual Trunk
- Media Gateway Controller
- SIP Line telephone corresponding to the B179 SIP Conference Phone

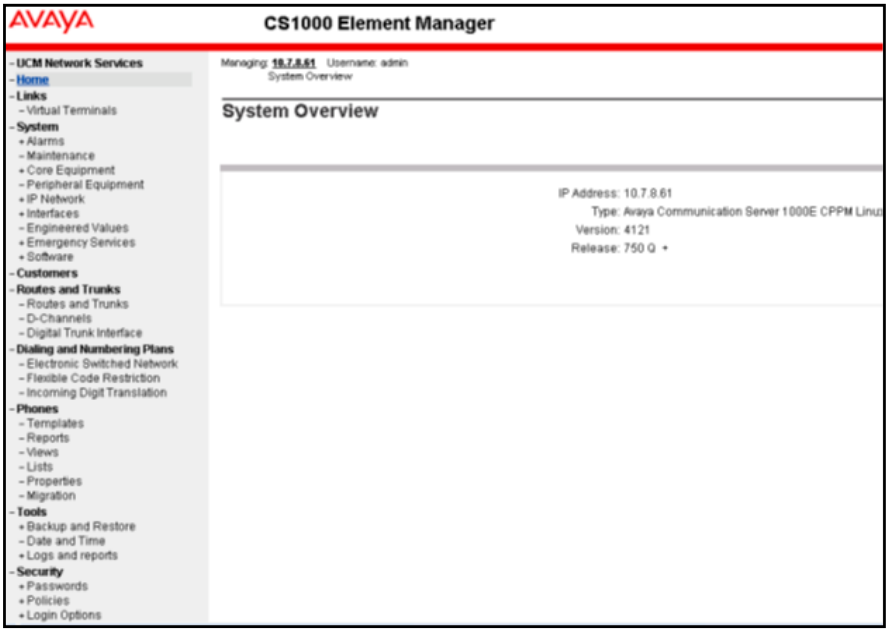
Procedure

1. Open Element Manager.

a. Log in to the web UI of Avaya Unified Communications Management (UCM) web interface by using the IP address of the CS1000 call server. The UCM page is displayed.



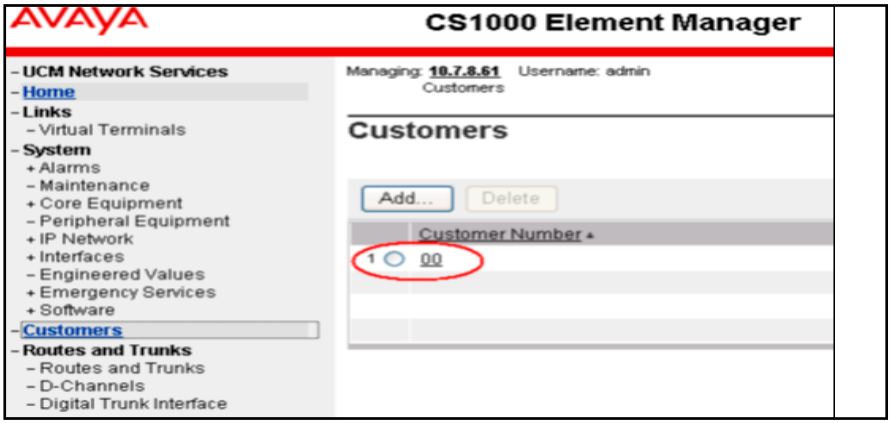
b. Click the Element Name corresponding to the Element Type CS1000. The System Overview page is displayed.



2. Enable SIP Line Service

a. In the left pane of the System Overview page, click Customers.

b. On the Customers page, click the Customer Number link.

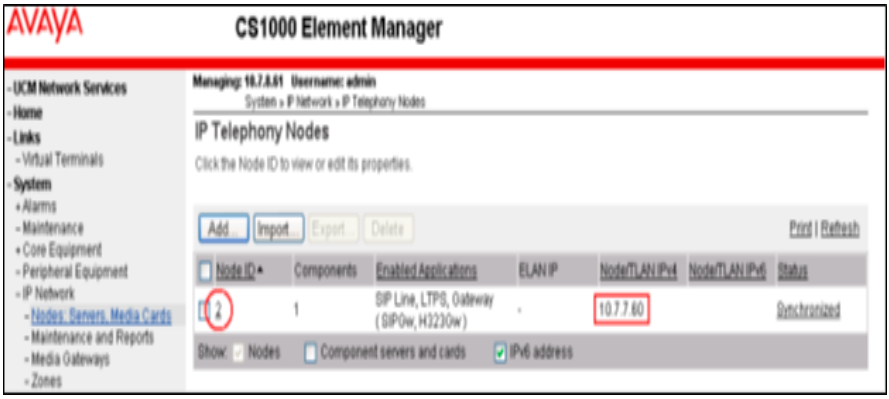


c. On the Customer Details page, click SIP Line Service.

d. Check the SIP Line Service checkbox, enter an appropriate User Agent DN prefix, and click Save.

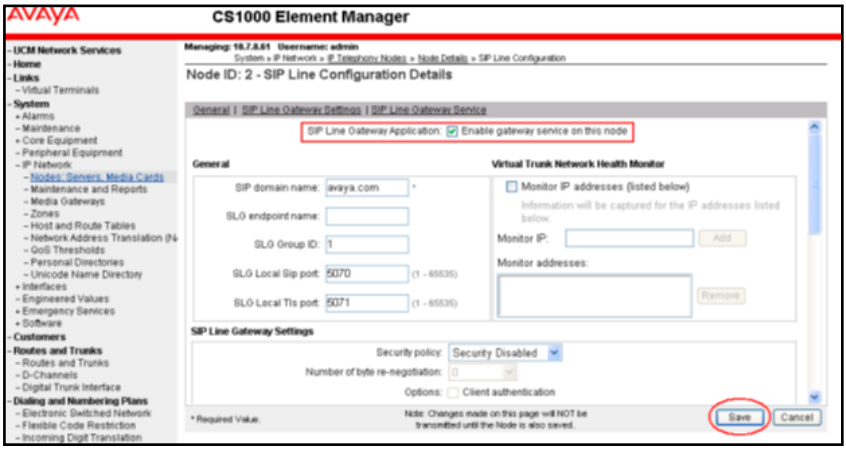
3. Enable SIP Line Service on Telephony Node.

a. On the Element Manager page, go to System > IP Network > Nodes: Servers, Media Cards. Note the IP address of the node, as it will be used in configuring the B179 later. Select the Node ID on which SIP Line service is to be enabled.



b. On the Node details page, under Applications, and click SIP Line.

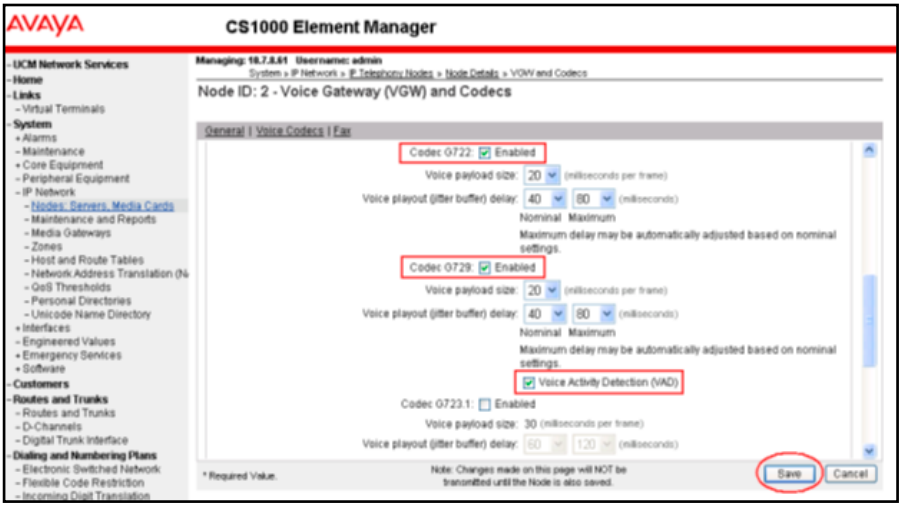
c. On the SIP Line Configuration Details page, select Enable gateway service on this node, and click Save.


AVAYA
CS1000 Element Manager
Node ID: 2 - SIP Line Configuration Details
General | SIP Line Gateway Settings | SIP Line Gateway Service
SIP Line Gateway Application: Enable gateway service on this node
General
Virtual Trunk Network Health Monitor
SIP domain name: avaya.com
SLG endpoint name:
SLG Group ID: 1
SLG Local Sip port: 5070 (1 - 65535)
SLG Local Tls port: 5071 (1 - 65535)
Monitor IP addresses (listed below)
Monitor IP:
Monitor addresses:
SIP Line Gateway Settings
Security policy: Security Disabled
Number of byte re-negotiation:
Options: Client authentication
* Required Value.
Note: Changes made on this page will NOT be transmitted until the Node is also saved.
Save
Cancel

d. On the Node Details page, click Voice Gateway (VGW) and Codecs, and select the required codecs.

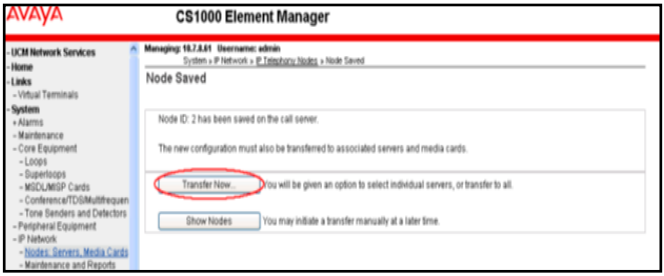
For G.722 and G.729 support, select Enabled next to Codec G.722 and Codec G.729.

For G.729 Annex B (silence suppression), select Voice Activity Detection (VAD). Ensure that the VAD setting on Element Manager is consistent with the VAD setting in the B179 phone.


AVAYA
CS1000 Element Manager
Node ID: 2 - Voice Gateway (VGW) and Codecs
General | Voice Codecs | Fax
Codec G722: Enabled
Voice payload size: 20 (milliseconds per frame)
Voice playout (jitter buffer) delay: 40 80 (milliseconds)
Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.
Codec G729: Enabled
Voice payload size: 20 (milliseconds per frame)
Voice playout (jitter buffer) delay: 40 80 (milliseconds)
Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.
Voice Activity Detection (VAD)
Codec G723.1: Enabled
Voice payload size: 30 (milliseconds per frame)
Voice playout (jitter buffer) delay: 60 120 (milliseconds)
* Required Value.
Note: Changes made on this page will NOT be transmitted until the Node is also saved.
Save
Cancel

e. Click Save to complete the node configuration. The Node Saved page is displayed.

f. On the Node Saved page, click Transfer Now.


AVAYA
CS1000 Element Manager
Node Saved
Node ID: 2 has been saved on the call server.
The new configuration must also be transferred to associated servers and media cards.
Transfer Now...
You will be given an option to select individual servers, or transfer to all.
Show Nodes
You may initiate a transfer manually at a later time.

After the transfer is complete, the Synchronize Configuration Files (Node ID <ID>) page is displayed.

g. Select the CS1000 call server and click Start Sync. The screen automatically refreshes till the synchronization is complete. After synchronization is complete, the status of the Synchronization Status field changes from Sync required to
Synchronized.

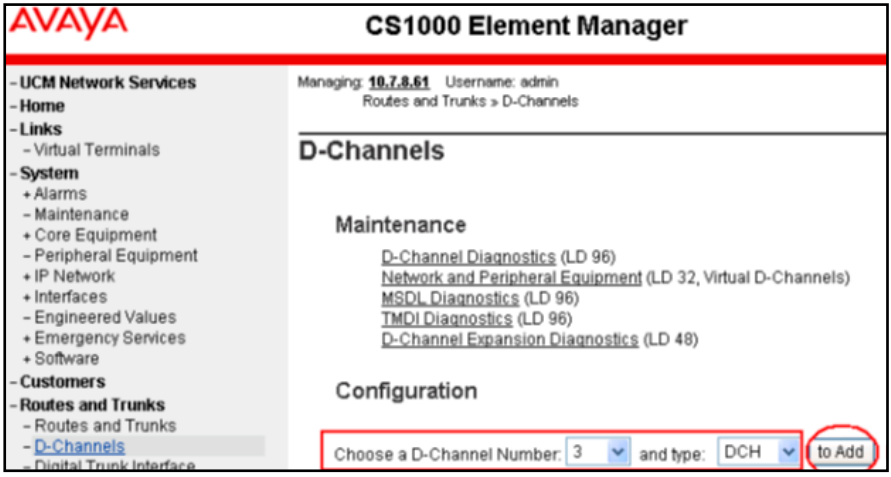
h. To use the new SIP Gateway settings, click Restart Applications.

4. Configure SIP Line D-Channel

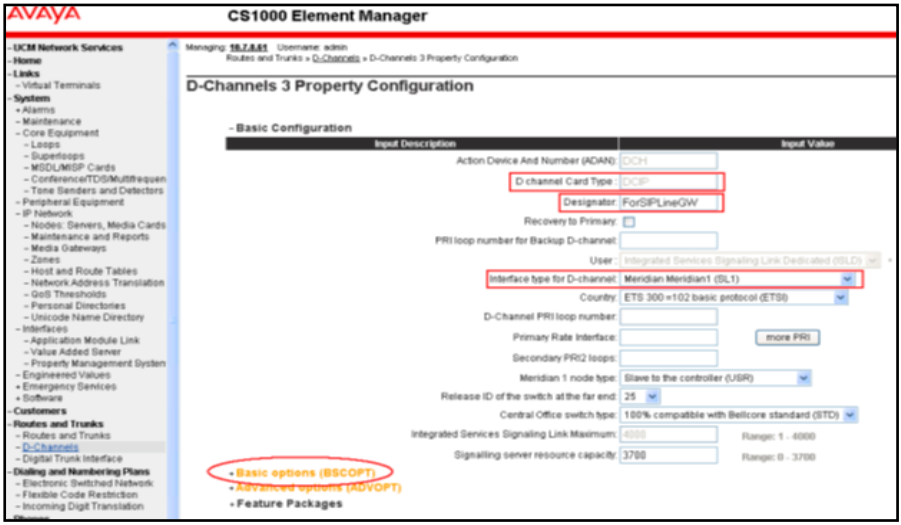
a. On the Element Manager page, click Routes and Trunks > D-Channels.

b. Choose a D-Channel number from the list, and select DCH as type.

c. Click to Add.

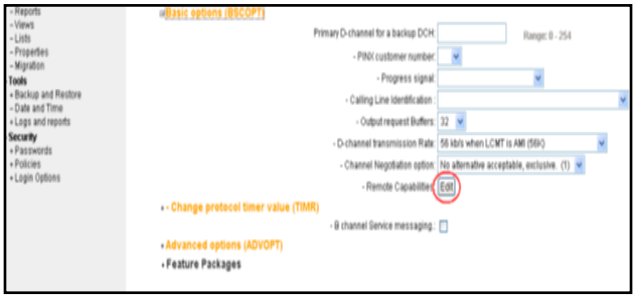


d. In the D-Channels Property Configuration page, select DCIP as the D-Channel Type, Meridian Meridian1(SL1) as the Interface type for D-channel, and a Designator. The remaining parameters can have default values.

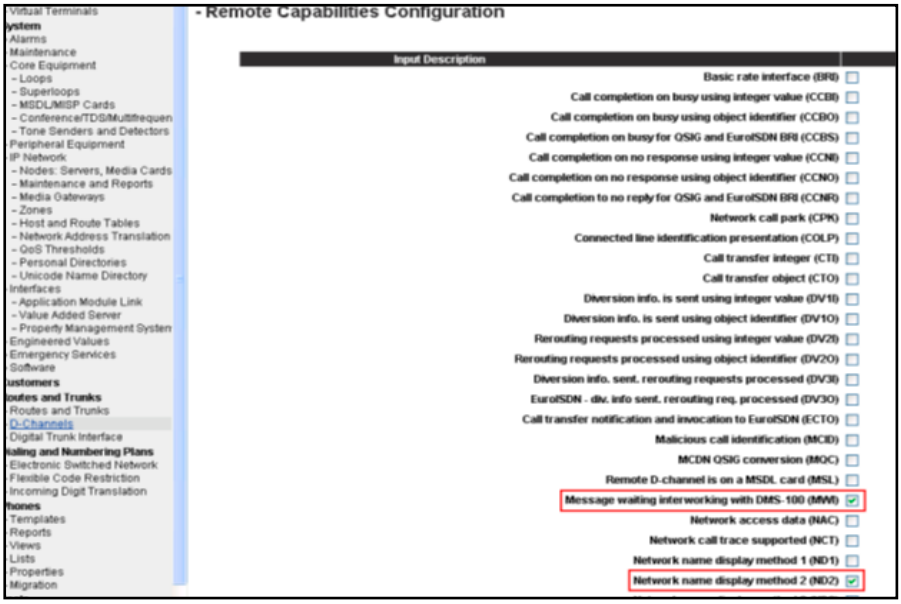


e. Click to expand the Basic options (BSCOPT) link.

f. Click the Edit button to configure Remote Capabilities.



g. On the Remote Capabilities page, select Message waiting interworking with DMS-100 (MWI) and Network name display method 2 (ND2).


- Remote Capabilities Configuration
Input Description
Basic rate interface (BRI)
Call completion on busy using integer value (CCBI)
Call completion on busy using object identifier (CCBO)
Call completion on busy for QSIG and EuroISDN BRI (CCBS)
Call completion on no response using integer value (CCNI)
Call completion on no response using object identifier (CCNO)
Call completion to no reply for QSIG and EuroISDN BRI (CCNR)
Network call park (CPK)
Connected line identification presentation (COLP)
Call transfer integer (CTI)
Call transfer object (CTO)
Diversion info. is sent using integer value (DV1I)
Diversion info. is sent using object identifier (DV1O)
Rerouting requests processed using integer value (DV2I)
Rerouting requests processed using object identifier (DV2O)
Diversion info. sent. rerouting requests processed (DV3I)
EuroISDN - div. info sent. rerouting req. processed (DV3O)
Call transfer notification and invocation to EuroISDN (ECTO)
Malicious call identification (MCID)
MCDN QSIG conversion (MQC)
Remote D-channel is on a MSDL card (MSL)
Message waiting interworking with DMS-100 (MWI)
Network access data (NAC)
Network call trace supported (NCT)
Network name display method 1 (ND1)
Network name display method 2 (ND2)

h. Click Return-Remote Capabilities.

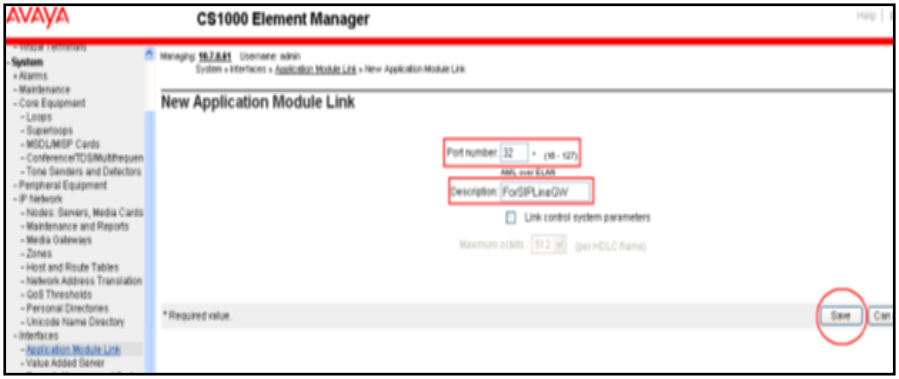
i. On the D-Channel Property Configuration page, click Submit.

5. Configure Application Module Link

a. On the Element Manager page, click System > Interfaces > Application Module Link > Add. The New Application Module Link page is displayed.

b. Enter the AML port number in the Port number text box. The SIP Line Service can use ports 32 through 127.

c. Enter a description, and click Save.



6. Configure Value Added Server (VAS)

a. On the Element Manager page, click System > Interfaces > Value Added Server > Add.

b. On the Add Value Added Server page, click Ethernet LAN Link.

c. On the Ethernet LAN Link page, enter a Value added server ID. For example, 064.

d. Select the AML number for Ethernet LAN Link.

e. Ensure that the Application Security check-box is not selected, and click Save.

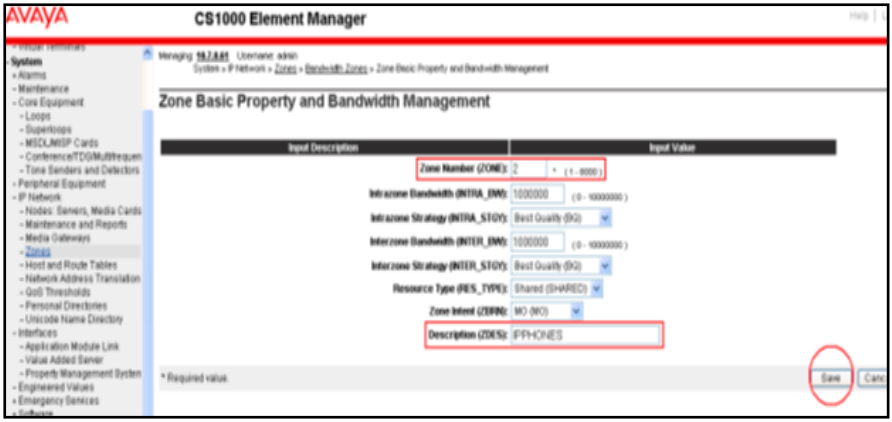
7. Configure Zone for SIP Phones

a. On the Element Manager page, click System > IP Network > Zones.

b. On the Zones page, click Bandwidth Zones.

c. On the Zone Basic Property and Bandwidth Management page, enter the Zone Number (ZONE) and description. The remaining parameters can have default values.

d. Click Save.



8. Configure SIP Line Route Data Block (RDB)

a. On the Element Manager page, click Routes and Trunks > Routes and Trunks.

b. Click the Add route button for the customer number.

c. On the Basic Configuration page, enter the values for the following parameters:

Route number (ROUT): Select a route number

Designator field for trunk (DES): Enter an appropriate name

Trunk type (TKTP): Select TIE trunk data block (TIE)

Incoming and outgoing trunk (ICOG): Select Incoming and Outgoing

(IAO)

Access code for the trunk route (ACOD): Enter the access code

The route is for a virtual trunk route (VTRK): Select the check-box

Zone for codec selection and bandwidth: Enter a zone (ZONE)

Node ID of signaling server of this route (NODE) Enter the node ID of the SIP Line Gateway

Protocol ID for the route (PCID): Select SIP Line (SIPL)

Integrated services digital network option (ISDN): Select the check-box

Mode of operation (MODE)

Select Route uses ISDN Signaling

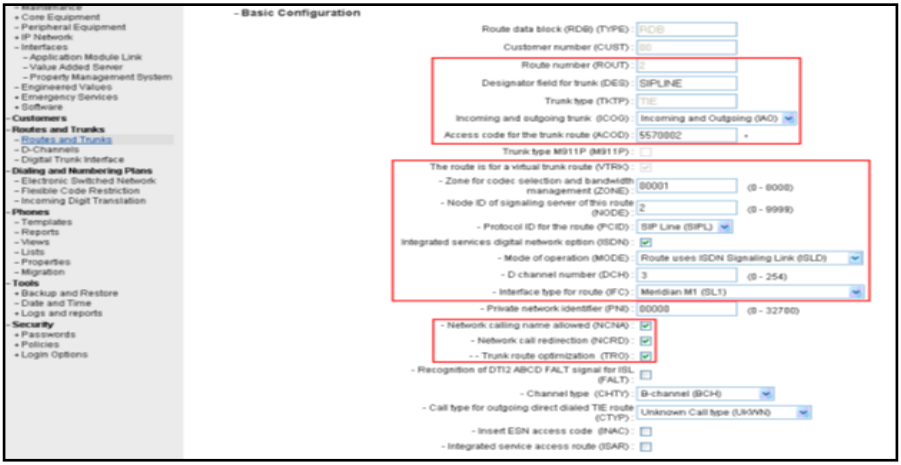
Link (ISLD)

D channel number (DCH) Enter the D-channel number

Interface type for route (IFC) Select Meridian M1 (SL1)

Network calling name allowed (NCNA) Check the box

Network call redirection (NCRD) Check the box


- Basic Configuration
Route data block (RDB) (TYPE):
Customer number (CUST):
Route number (ROUT):
Designator field for trunk (DES):
Trunk type (TKTP):
Incoming and outgoing trunk (ICOG):
Access code for the trunk route (ACOD):
Trunk type M911P (M911P):
The route is for a virtual trunk route (VTRK):
- Zone for codec selection and bandwidth management (ZONE):
- Node ID of signaling server of this route (NODE):
- Protocol ID for the route (PCID):
Integrated services digital network option (ISDN):
- Mode of operation (MODE):
- D channel number (DCH):
- Interface type for route (IFC):
- Private network identifier (PNI):
- Network calling name allowed (NCNA):
- Network call redirection (NCRD):
- Trunk route optimization (TRO):
- Recognition of DTI2 ABCD FALT signal for ISL (FALT):
- Channel type (CHTY):
- Call type for outgoing direct dialed TIE route (CTYP):
- Insert EGN access code (INAC):
- Integrated service access route (ISAR):

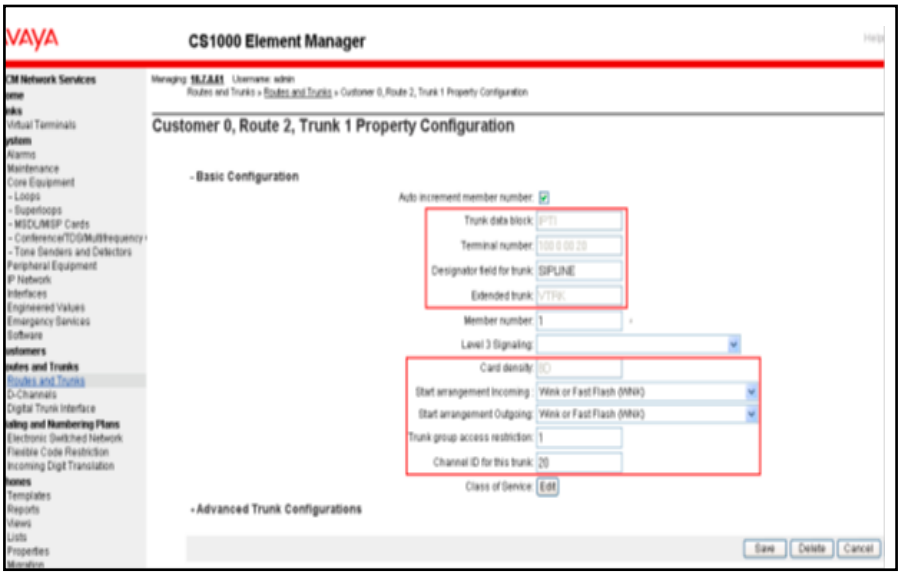
9. Configure SIP Line Virtual Trunk

a. On the Element Manager page, click Routes and Trunks > Routes and Trunks.

b. To add new trunk members to the new SIP line route, click Add trunk.

c. On the Trunk Property configuration page, set the following parameters to the values specified. The remaining parameters can have default values.

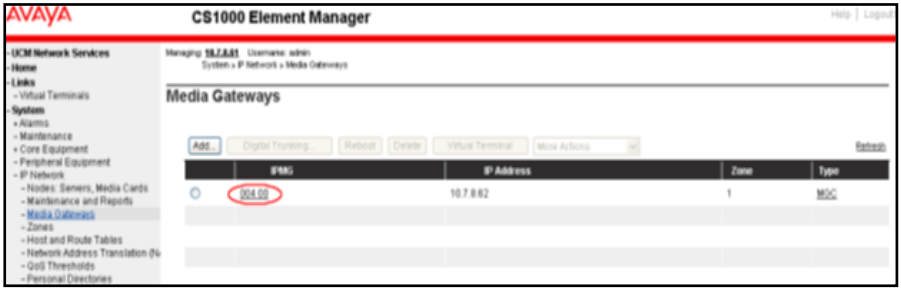
d. Click Save.



10. Configure Media Gateway Controller

a. On the Element Manager page, click IP Network > Media Gateways.

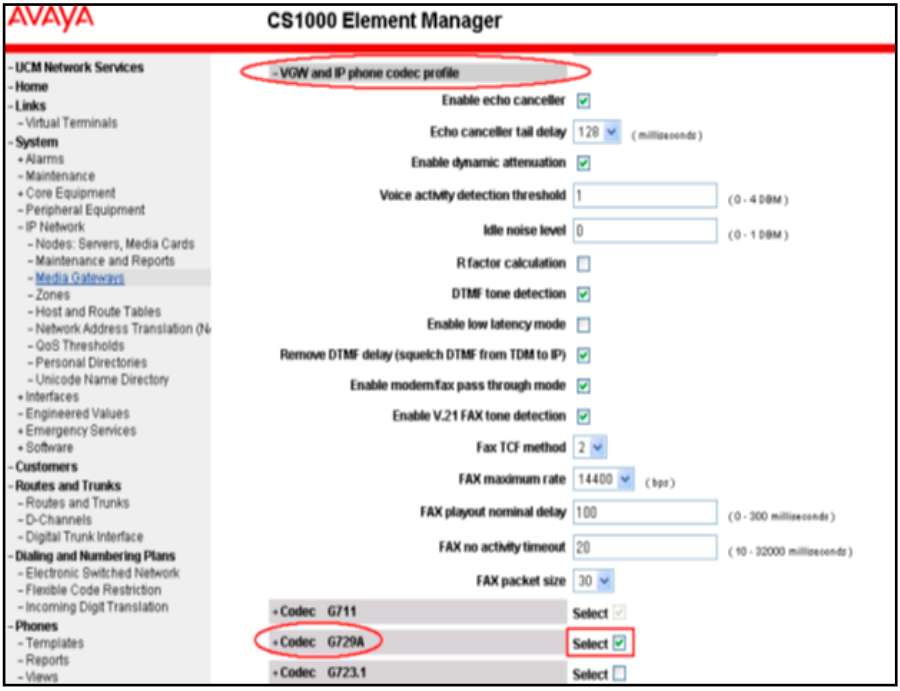
b. In the IPMG column, click the option that supports the digital and analog phones in the communication system.



c. On the IPMG Property Configuration page, click Next.

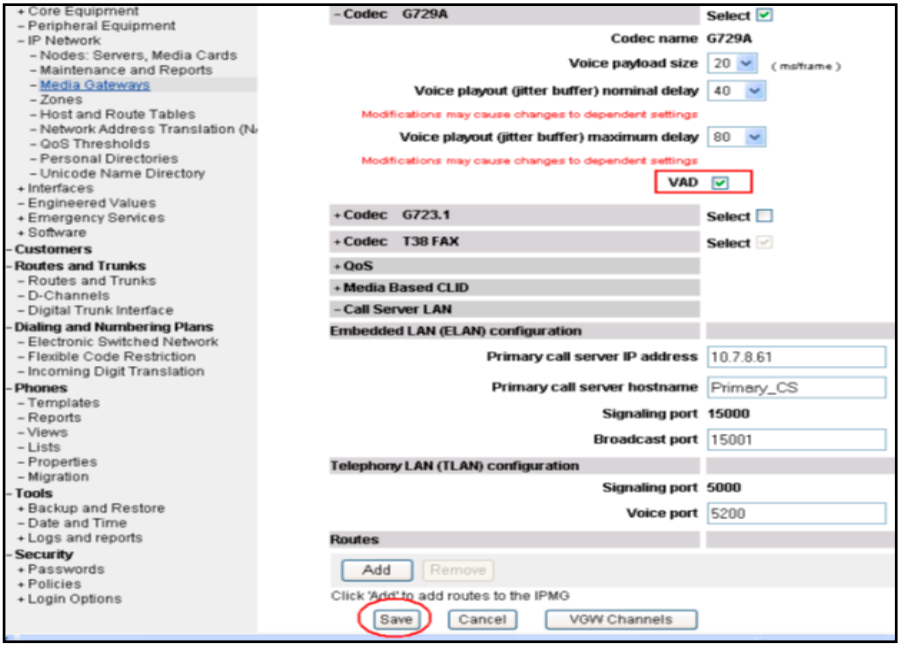
d. Click to expand the VGW and IP phone codec profile section.

e. Select to expand the Codec G729A option.



f. For G.729 Annex B (silence suppression), select Voice Activity Detection (VAD). Ensure that the VAD setting on Element Manager is consistent with the VAD setting in the B179 phone.

g. Click Save.

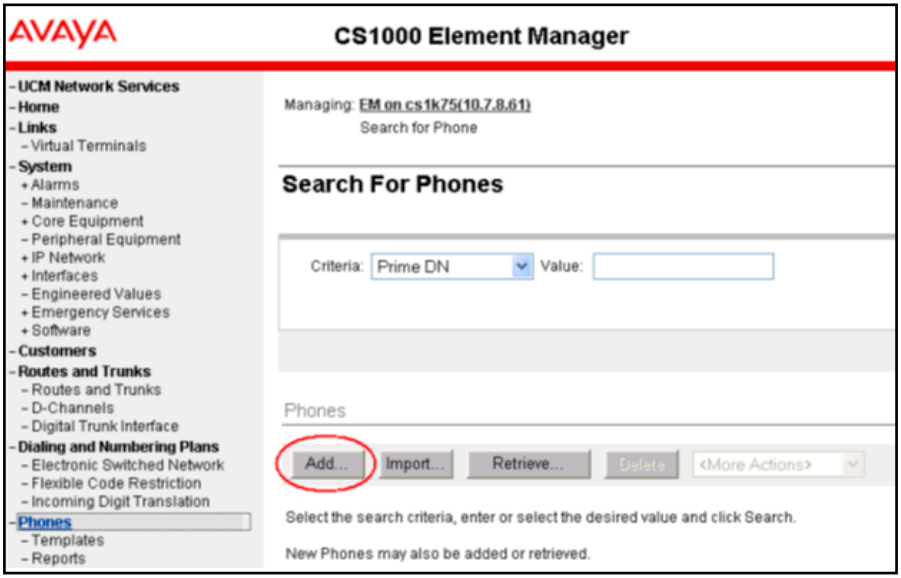


h. On the Media Gateway page, select the configured IPMG and click Reboot.

11. Configure SIP Line Telephone

a. On the Element Manager page, click Phones.

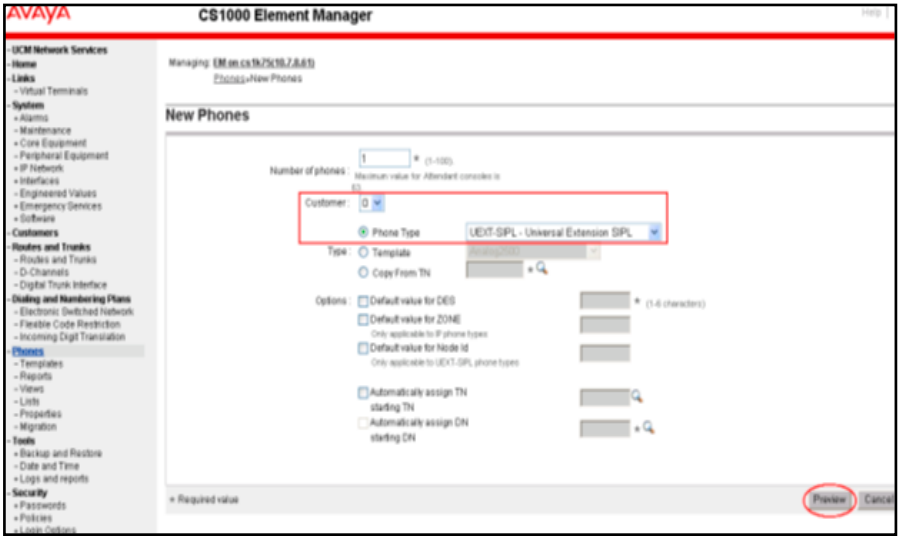
b. On the Search for Phones page, click Add.



c. On the New Phones page, select the Customer.

d. Select the Phone Type radio button, and from the list of options, select UEXT-SIPL – Universal Extension SIPL.

e. Click Preview.



APPENDIX C: USING CERTIFICATES

Follow these steps to authenticate the Avaya B179 SIP Conference Phones using TLS/SIPS and EAP-TLS:

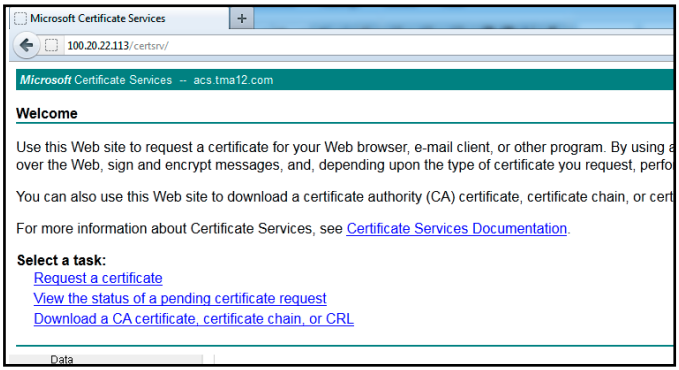
Download the root certificate from the Certificate Server.
Create the server certificate from the Certificate Server.
Generate the private key.
Convert the certificates and private key to .PEM format.
Import the .PEM files to the B179 Phone.

ⓘ For information about using EJBCA certificates on Avaya Aura® System Manager, see the Administering Avaya Aura® System Manager document.

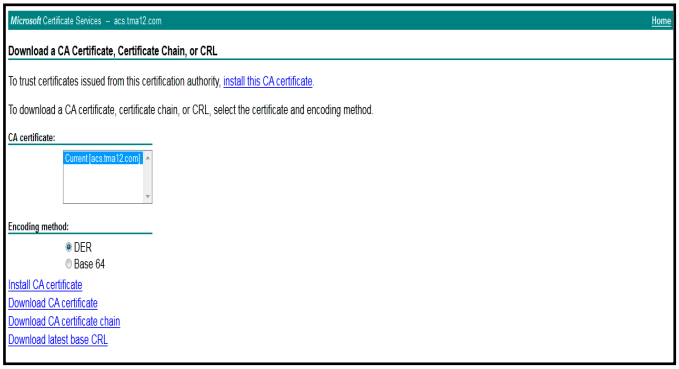
Procedure

1. Download the root certificate

In the Microsoft Server Certification Authority page, click Download a CA certificate, certificate chain, or CRL.


Microsoft Certificate Services
100.20.22.113/certsrv/
Microsoft Certificate Services -- acs.tma12.com
Welcome
Use this Web site to request a certificate for your Web browser, e-mail client, or other program. By using a
over the Web, sign and encrypt messages, and, depending upon the type of certificate you request, perfo
You can also use this Web site to download a certificate authority (CA) certificate, certificate chain, or cert
For more information about Certificate Services, see Certificate Services Documentation.
Select a task:
Request a certificate
View the status of a pending certificate request
Download a CA certificate, certificate chain, or CRL
Data

To download the root certificate for Avaya B179, click Download CA certificate.

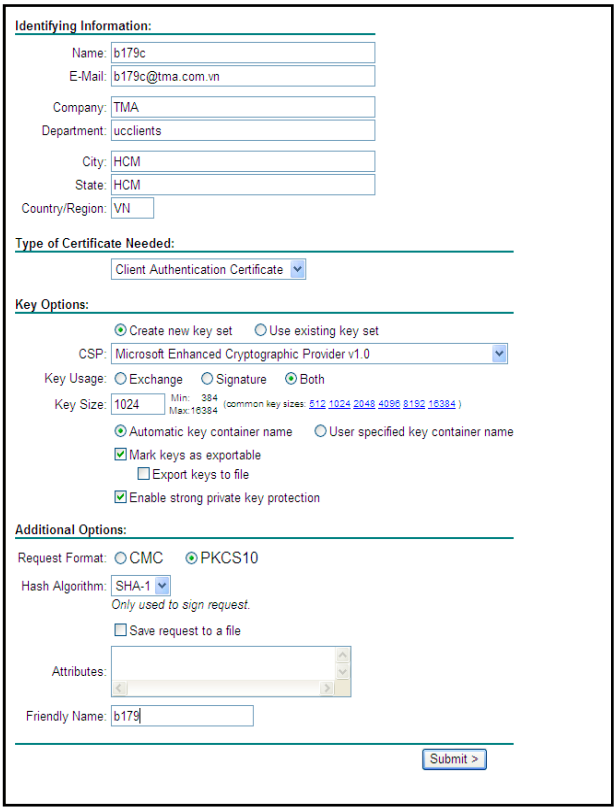

Microsoft Certificate Services -- acs.tma12.com
Home
Download a CA Certificate, Certificate Chain, or CRL
To trust certificates issued from this certification authority, install this CA certificate.
To download a CA certificate, certificate chain, or CRL, select the certificate and encoding method.
CA certificate:
Current [acs.tma12.com]
Encoding method:
DER
Base 64
Install CA certificate
Download CA certificate
Download CA certificate chain
Download latest base CRL

2. Get the server certificate

a. On the Microsoft Server Certification Authority page, click Request a certificate.

b. On the Request a certificate page, click advanced certificate request.

c. Enter the information required to create the certificate, and click Submit.



d. The certificate is saved to the location specified while setting up the CA.

3. Install the certificate

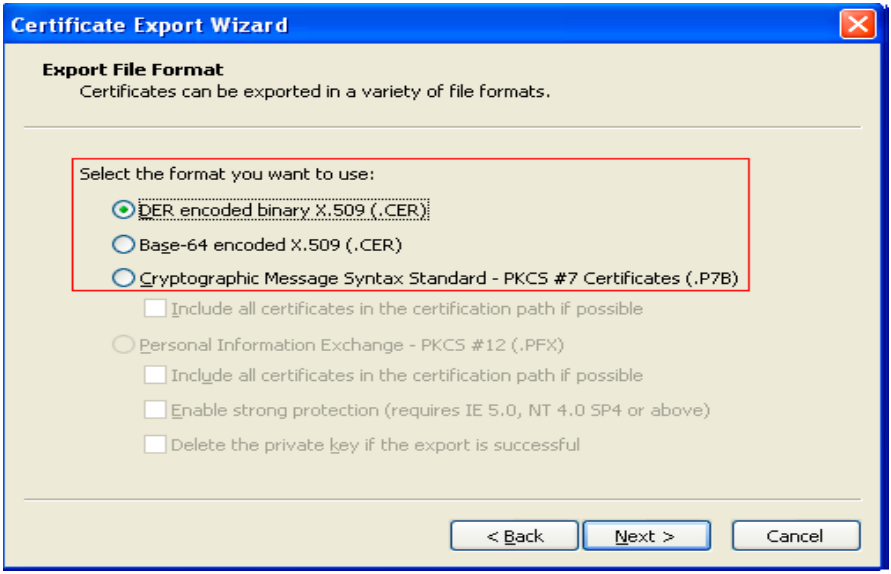
a. Import the certificate to the web browser.

4. Export private key

a. Go to Internet Options > Content > Certificates, select the certificate you installed, and click Export.

b. Select Yes, export the private key, and click Next.

c. Select the format in which you want to export the certificate file, and click Next.


Certificate Export Wizard
Export File Format
Certificates can be exported in a variety of file formats.
Select the format you want to use:
DER encoded binary X.509 (.CER)
Base-64 encoded X.509 (.CER)
Cryptographic Message Syntax Standard - PKCS #7 Certificates (.P7B)
Include all certificates in the certification path if possible
Personal Information Exchange - PKCS #12 (.PFX)
Include all certificates in the certification path if possible
Enable strong protection (requires IE 5.0, NT 4.0 SP4 or above)
Delete the private key if the export is successful
< Back
Next >
Cancel

d. Specify the file name and browse to the location to where the certificate must be exported.

6. Convert the certificates to .PEM format

The B179 phone supports certificates in the .PEM format only. You must convert the certificates and private keys to .PEM before using in the B179 phones.

a. Use the following Openssl commands to convert the files:

From .DER to .PEM

```
openssl x509 -inform der -in certificate.cer -out certificate.pem
```

From .PFX to .PEM

```
openssl pkcs12 -in certificate.pfx -out certificate.cer -nodes
```

Next steps

Browse to the .PEM files from the B179 web UI to use EAP TLS mode of authentication.